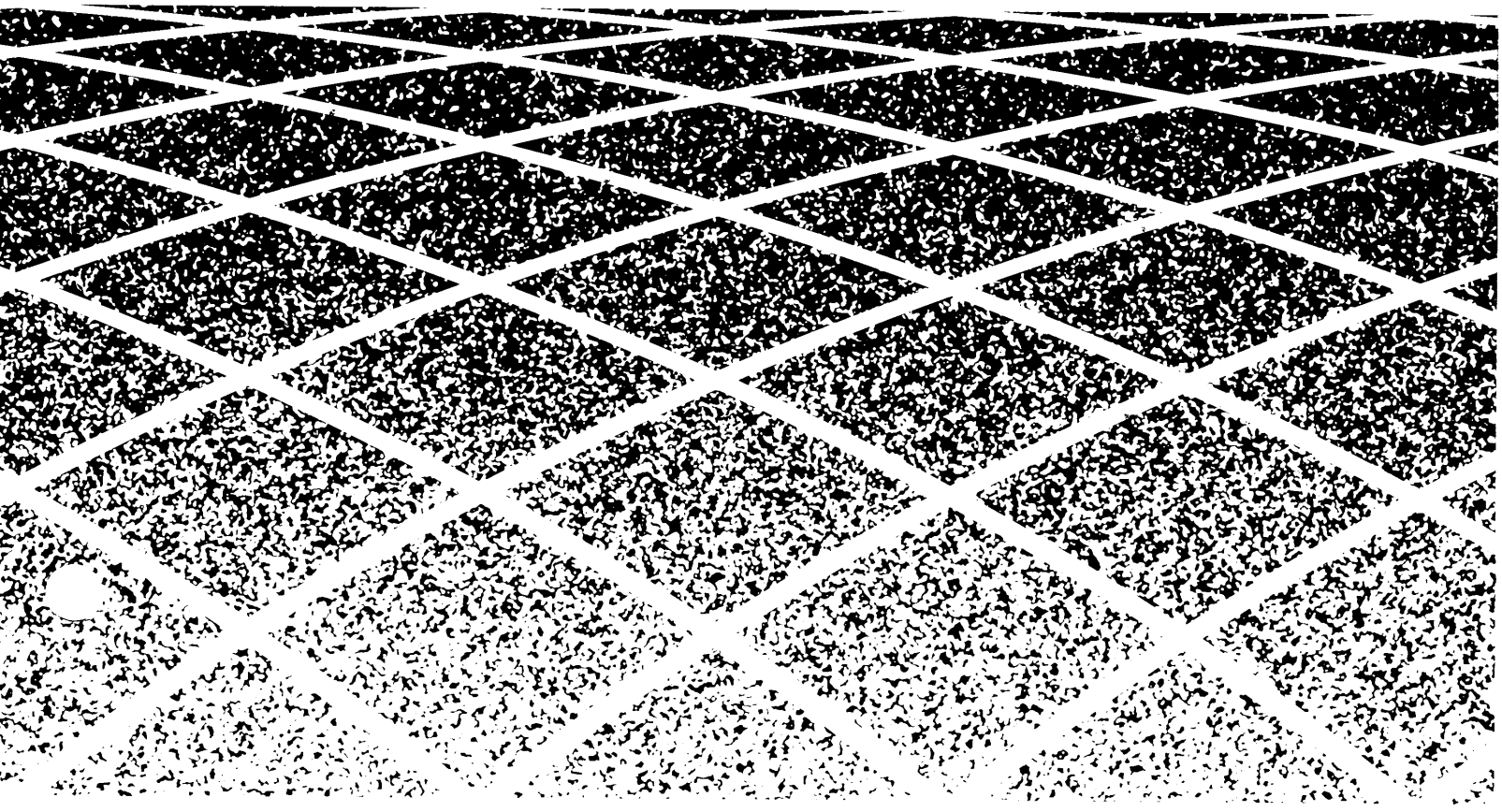
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Issue 1, June 1988

AT&T SYSTEM 25

R2V1
REFERENCE MANUAL



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Indianapolis, Indiana 42619

Every effort was made to ensure that the information in this document was complete and accurate at the time of printing. However, this information is subject to change. This document will be reissued periodically to incorporate changes.

R2V1 Reference Manual
Prepared by System 25
Document Development Group and the
AT&T Documentation
Management Organization

FCC NOTIFICATION AND REPAIR INFORMATION

AT&T SYSTEM 25

This telephone equipment is registered with the Federal Communications Commission (FCC) in accordance with Part 68 of its Rules. In compliance with the Rules, be advised of the following:

MEANS OF CONNECTION

Connection of this telephone equipment to the nationwide telecommunications network shall be through a standard network interface USOC RJ21X jack. Connection to private line network channels requires USOC RJ2GX jack for tie lines or USOC RJ21X jack for off-premises station lines. These can be ordered from your telephone company.

NOTIFICATION TO THE TELEPHONE COMPANY

If the system is to be connected to off-premises stations (OPSs), you must notify the telephone company of the OPS class of service, OL13C, and the service order code, 9.0F.

Upon the request of the telephone company, inform them of the following:

- The Public Switched Network “lines” and the Private “lines” to which you will connect the telephone equipment.
- The telephone equipment’s “registration number” and “ringer equivalence number” (REN) from the label on the equipment.
- For private line connections, provide the facility interface code, TL31M for tie lines. You must also specify the service order code, 9.0F.
- The quantities and USOC numbers of the jacks required.
- For each jack, provide the sequence in which lines are to be connected; the type lines and the facility interface code and the ringer equivalence number by position, when applicable.

This telephone equipment should not be used on coin telephone lines. Connection to party line service is subject to state tariffs.

REPAIR INSTRUCTIONS

If you experience trouble with this telephone equipment, contact the AT&T National Service Assistance Center on 1-800-628-2888. The telephone company may ask that you disconnect this equipment from the network until the problem has been corrected or until you are sure that this equipment is not malfunctioning.

RIGHTS OF THE TELEPHONE COMPANY

If your telephone equipment causes harm to the telephone network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice isn't practical, you will be notified as soon as possible. You will be informed of your right to file a complaint with the FCC.

Your telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted telephone service.

HEARING AID COMPATIBILITY

The voice terminals described in this manual are compatible with inductively coupled hearing aids as prescribed by the FCC.

FCC REGISTRATION INFORMATION	
Registration Number	AS593M-71565-MF-E
Ringer Equivalence	0.5A
Network Interface	RJ21X or RJ2GX

PRIVATE LINE SERVICE	
Service Order Code	9.0F
Facility Interface Code	
● Tie Lines	TL31M
● Off-Premises Stations	OL13C

FCC WARNING STATEMENT

Federal Communications Commission (FCC) Rules require that you be notified of the following:

- This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause interference to radio communications.
- It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment.
- Operation of this equipment in a residential area is likely to cause interference in which case the user at his or her own expense will be required to take whatever measures may be required to correct the interference.

DANGER

The AT&T System 25 cabinets are not user serviceable. Some voltages inside the cabinets are hazardous. This equipment is to be serviced only by qualified technicians.

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Section 2—Features and Services

Section 3—Functional Description

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Section 5—Technical Specifications

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OVERVIEW

Introduction

This reference manual provides general technical information on AT&T System 25 (System 25). It includes a description of the system, its hardware and software, features and services, environmental requirements, and technical specifications. This manual is intended to serve as an overall technical reference for System 25.

This manual is released specifically to cover Release 2 Version 1 (R2V1) of System 25. It does not contain information that applies only to the earlier releases of System 25.

Organization

This manual is divided into 10 sections. The remaining sections are as follows:

- SECTION 2—FEATURES AND SERVICES
- SECTION 3—FUNCTIONAL DESCRIPTION
- SECTION 4—HARDWARE DESCRIPTION
- SECTION 5—TECHNICAL SPECIFICATIONS
- SECTION 6—ENVIRONMENTAL REQUIREMENTS
- SECTION 7—PARTS INFORMATION
- SECTION 8—REFERENCE DOCUMENTATION
- SECTION 9—GLOSSARY
- SECTION 10—INDEX.

System 25 Description

System 25 is an advanced digital switching system that integrates voice and data communications. (See block diagram in Figure 1-1.) It not only provides the features of a state-of-the-art Private Branch Exchange (PBX), but goes a step further by allowing data to be switched point-to-point without first being converted to analog format. This capability can be used to set up connections between data terminals, word processors, personal computers, and host computers. System 25 uses intelligent port circuits equipped with distributed network processor elements to provide (essentially) nonblocking voice and data switching.

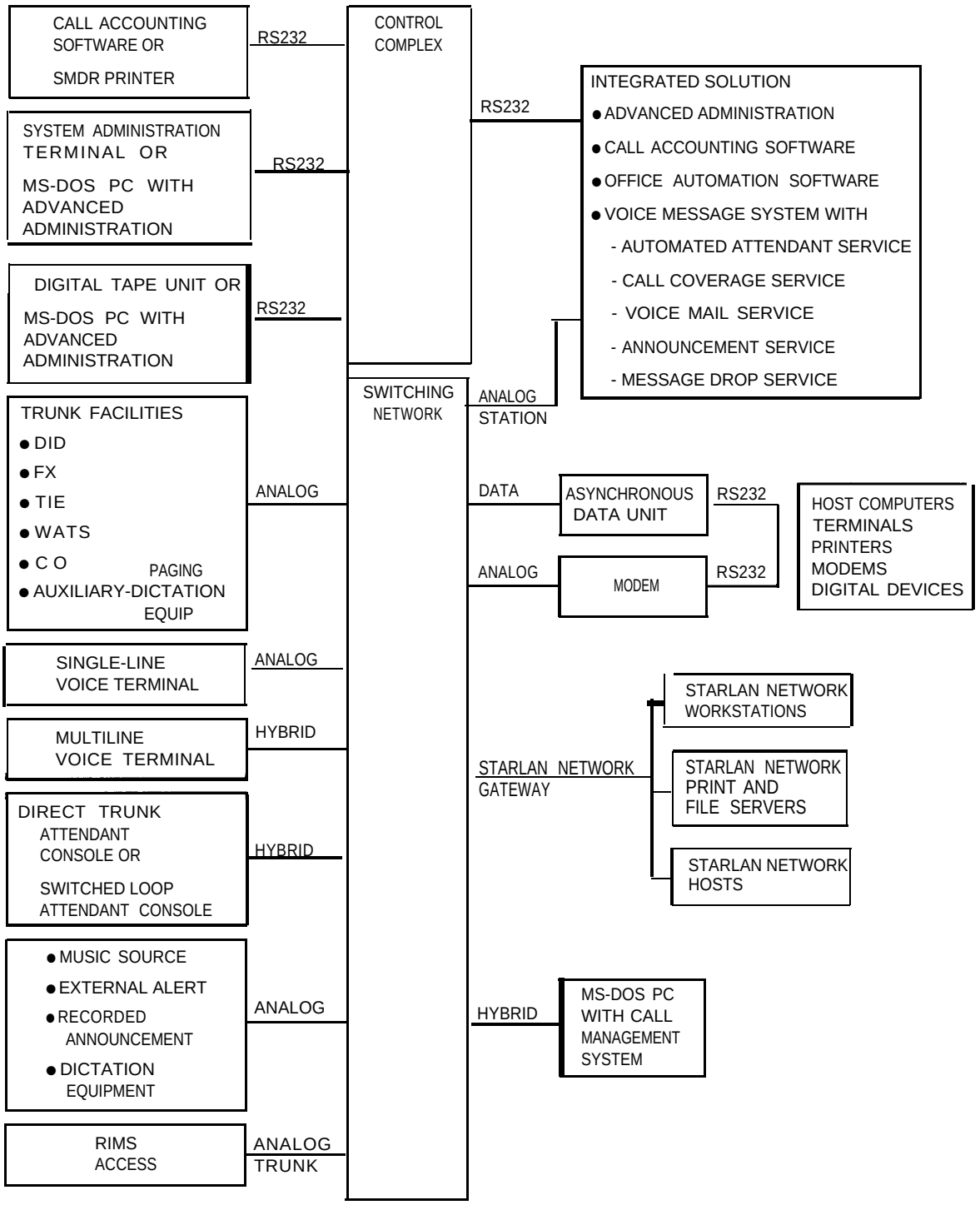


Figure 1-1. System 25 Block Diagram

Voice communications features combine traditional telephone features, such as Transfer and Hold, with advanced features, such as Individual and Group Coverage, Hands-Free Answer On Intercom, and Speed Dialing (see Section 2, “Features and Services”).

Data communications features provide switched data connections supporting transmission of voice and data over Premises Distribution System wiring. Data connections can be made between two digital data modules (asynchronous data units), between two analog modems, or between an analog modem and a digital data module. System 25 also provides access to STARLAN NETWORKs (Release 2 of STARLAN only). The system has data modules that provide an RS-232 interface for full duplex, asynchronous, transmission of data up to 19,200 bps, and an integrated 212A-compatible modem pool resource.

System 25’s Integrated Solution offers customers a unique package of integrated call management, switch management, and office automation applications. The Integrated Solution is a set of application programs that run on a Master Controller (UNIX® PC). The applications include Advanced Administration Software (AAS), which permits customers to administer System 25’s features themselves; Call Accounting System (CAS); and an integrated Voice Message System (VMS) that provides call coverage, leave word calling, automated attendant, and voice mail services. In addition, a number of generic office automation applications (word processing, data base management, and spreadsheet) are also available for the Integrated Solution; these applications may be run simultaneously with the VMS and CAS applications.

System 25 supports the following:

- Trunk and Network Facilities— Dual Tone Multifrequency (DTMF) and Dial Pulse Signaling on incoming and outgoing trunks (dial pulse only on DID trunks).
  - — Loop Start (LS)
  - — Ground Start (GS) (Strongly Preferred over Loop Start in most installations)
  - — Tie Trunks (Type I and Type I Compatible E&M, Type V Simplex)
  - — Direct Inward Dialing (DID)
- Voice Terminals — Single-Line Touch-Tone, Single-Line Rotary, MET, and 7300H Series Multiline Sets from the MERLIN® Communications System.
- Data Facilities
  - — Digital Data End Points — RS-232 Interfaces via Asynchronous Data Units
  - — Analog Data End Points — Tip/Ring-Type Modem Interfaces
  - — STARLAN NETWORK Access (Release 2 of STARLAN only).

- Networking Capability
  - — Remote Access
  - — Tie Trunks
  - — Tandem Trunking
  - — Endpoint in Electronic Tandem Network (Tributary only, not Satellite)
  - — Endpoint of Enhanced Private Switched Communications Services (EPSCS)
  - — Endpoint of Tandem Tie Trunk Network (TTTN)
  - — Endpoint of Common Control Switching Arrangement (CCSA).

Call Handling Capabilities

System 25 can be arranged as a stand-alone system or can be part of a private network. The system provides 256 ports to support the following:

- 115 simultaneous two-party conversations
- Traffic Handling Capacity of 4140 CCS (Trunking Limited)
- Busy Hour Call Capacity of 2500 calls (DTMF Register Limited)
- Up to 104 trunk ports including Central Office (CO), DID, Tie, Foreign Exchange (FX), Wide Area Telecommunications Service (WATS), and 800 Service
- An Auxiliary Trunk interface for paging and dictation systems
- Up to 240 ports that support a combination of the following:
  - — Up to 200 ports for voice terminals and auxiliary feature port equipment.
  - — Up to 104 data ports providing RS-232 connections to data terminals, personal or multiport computers.

Refer to Hardware and Software Parameters as provided in "Technical Specifications" (Section 5) for detailed specifications.

Safety

System 25 meets all requirements found in Underwriters Laboratories Standard for Telephone Equipment (1459).

Business Communications Needs

The remainder of this section describes how System 25's features may be used to satisfy a customer's communications needs. This material may be thought of as the reverse of the "Features and Services" in Section 2.

The business communications capabilities of the majority of small businesses with more than 30 phones are provided by a PBX. System 25 is a PBX designed to meet the business communications needs of customers in the 30 to 150 station range.

The communications needs of most business customers may be broken down into five basic categories. Customer experience has shown that a PBX needs to provide—

- Prompt handling of incoming calls to maximize revenue opportunities and client satisfaction,
- Ease of access to and cost control of outgoing calls over public network and private facilities,
- Easy movement of calls between on-premises phones and between on-premises and off-premises phones,
- Sharing of data between PCs and/or host computers and data terminals, and
- Growth and rearrangement of facilities.

The following pages outline System 25's outstanding ability to provide these services.

Incoming Business Communications

Successful call termination is the key to capturing all incoming communications associated with revenue issues, client inquiries, decision data, etc. Call termination involves identifying the called party and routing the call to a primary or secondary answering position. System 25 provides powerful tools for both call screening and call termination.

- Attendant Consoles allow one or two attendants to answer, screen, and steer incoming calls using either Direct Trunk or Switched Loop operation. With attendant operation, incoming calls can be screened and extended to the appropriate party for resolution or forwarded to alternate locations, and messages can be taken for absent clients. Calls may arrive over any of the network facilities described in later sections of these notes.
- System 25's Integrated Solution can provide Automated Attendant service, either reducing the volume of calls your attendant needs to handle or providing off-hour attendant service.

- Direct Inward Dialing allows incoming callers to reach specific individuals or facilities without attendant assistance. This allows specific numbers to be advertised for direct customer access to brokers, emergency services, etc., over a shared pool of DID trunks.

- The Call Management System provides Automatic Call Distribution (ACD) service and associated call traffic and agent performance reports.

- Direct Group Calling (DGC) allows incoming calls to be directed to a specific group of stations. Calls to a DGC group hunt for an idle station in a circular manner, starting at the station following the last one called. If all group members are busy, calls are queued and can be sent to a delay announcement. A DGC group can terminate calls to sales, services, computer, announcement, etc., over either ordinary CO trunks or DID trunks.

- Personal Lines provide dedicated outside lines for multiline voice terminal users and are accessed via a dedicated button for both incoming and outgoing service. Up to 16 terminals may share a Personal Line with up to 4 parties simultaneously off-hook. A personal line provides direct access to brokers, emergency service, etc., over a dedicated loop start or ground start trunk.

- Call Waiting lets users know that they have another incoming call and helps avoid missing important calls.

- Remote Access allows employees to use the services and facilities of System 25 from home or when they are on the road. Barrier codes prevent unauthorized access.

Frequently, the called party is not available to handle an incoming call. System 25 provides a number of methods for redirecting incoming calls to alternate resources.

- Coverage allows calls that are not answered within a specified number of rings to be redirected to an individual covering station and/or a group of covering stations. This is especially useful for Boss-Secretary arrangements, staff backup, and message service. This feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special features, such as the Send All Calls feature which routes a user's calls directly to covering station(s), accommodate the day-to-day variations that occur in an employee's work schedule.

- Following and Forwarding allow users who are away from their normal locations to receive their calls at other phones inside the system or (Forwarding only) outside the system. This feature supports roving personnel and shared office space for company staff.

- The Integrated Solution can provide call coverage service, along with integrated voice mail and Leave Word Calling.

- The Bridging feature permits calls on a user's System Access buttons to be answered at another station on Bridged Access buttons.

- Station Hunting provides automatic redirection of incoming calls to an idle member of a hunt group when the called party is busy.

- Pickup allows a user to answer a call ringing at another terminal. Directed Pickup allows a user to answer a call ringing at any terminal by dialing the pickup code and the Personal Dial Code (PDC) of the ringing station. Group Pickup permits calls to any other terminal in the pickup group to be answered by dialing the group call pickup code. With Pickup, users do not have to leave their phone to answer other's calls. This feature is especially useful for local coverage in group offices not supported by secretarial service and equipped with economical single-line phones.

When alternate resources are not available to handle an incoming call, System 25 provides for attendant handling of the call utilizing camp-on, redirection, and/or message service.

- Camp-On allows the attendant to extend an outside call to a busy station. A burst of tone is heard at the called station to notify the user of the camped-on call. The caller is placed on hold and hears music-on-hold, if available. When the user hangs up, the camped-on call begins ringing immediately. The Return Coverage on Busy feature returns unanswered camped-on calls to the attendant for service after a specified interval.

- Return Coverage on Don't Answer returns unanswered attendant-extended calls for additional service (redirection/messaging).

- Messaging Service supports activation of a light-emitting diode (LED) at the called station to indicate that the attendant, message desk, or another station has a message for the user.

Special arrangements are needed to handle incoming calls during periods when the normal staff is not available, for example, at night and on weekends. System 25's Night Service feature allows on-duty personnel to answer incoming attendant-seeking calls when the attendant is not on duty. Directed Night Service redirects incoming attendant-seeking calls to designated voice terminals, such as a guard desk or coverage position. Trunk-Answer-From-Any-Station (TAAS) Night Service allows users to answer incoming calls from any station by dialing the Night Service access code. Night personnel can be alerted by a night bell.

Outgoing Business Communications

One of the key functions of a customer premises communications system is to provide easy access to the most cost effective network facilities for outgoing calls. The system needs to be capable of steering calls based on cost, and must also keep records of incoming and outgoing calls and associated costs. Building on its ground start trunk capability, System 25 features control costs and record usage as follows.

- Call Restrictions allow the manager to restrict users from making certain types of calls. Restriction is administered through outward restriction, toll call restriction, and facility access restriction.

- Automatic Route Selection provides manager defined routing of calls over the telecommunications network based on preferred routes (normally the least expensive route available at the time the call is placed) with capacity for multiple common carriers and routing through tandem switch points. The user dials a standard Direct Distance Dialing (DDD) number and the system selects the call route.

- Station Message Detail Recording (SMDR) generates detailed call information on all incoming and outgoing calls and sends this information to an output device (PC or printer).

- Call Accounting Systems provide multiple types of customer reports on communication costs and usage.

- Account Code Entry allows a user to associate calls with an account code for charge-back purposes. This feature can be administrated (on a per-station basis) to force the entry of the required codes before outgoing calls can be made.

Ease of access to multiple types of network facilities (provided for minimum cost) is managed by the following features.

- Automatic Route Selection (ARS) allows the customer to dial a standard DDD number. ARS selects the preferred route and does any number conversions required for the facilities selected.

- System 25's Virtual Facility feature provides convenient and inexpensive access to Other Common Carriers (OCCs). This feature provides access to OCC facilities over a user specified *physical* facility; dedicated OCC trunks are not needed. Local OCC access numbers and account codes are automatically added by System 25. System 25's Virtual Facility feature is fully integrated with its ARS, Toll Restriction, and SMDR/CAS features.

- Callback Queuing provides a simple way to complete calls to busy trunk pools without having to manually repeat the calling procedures. Such calls are put into a queue; when the busy facility is available, the originator is alerted and the call is completed.

- Last Number Dialed automatically saves the last number dialed and allows the user to retry the number without redialing (multiline voice terminals only).

- Callback Queuing puts a call made to a busy facility into a queue, notifies the calling user when the facility becomes available to receive the call, and completes the call.

- Repertory Dialing allows multiline voice terminal users to store a telephone number or account and associate that number with a button on their voice terminal. Pressing a Repertory Dialing button is equivalent to dialing the stored number (one-touch dialing).

- System Speed Dialing allows all users to dial 90 selected numbers using 3-digit codes. Users can also program up to 20 Personal Speed Dialing Numbers, which are accessible only from their terminals. System Speed Dialing can be used by the system administrator to hide business account codes from users.

- Pooled Facility-Dial/Direct Access allows both multiline and single-line voice terminal users to access a common pool of trunks for outgoing calls by dialing a facility access code, or, on multiline voice terminals, by pressing a button. This grouping provides resource pooling, which results in better service with a given number of trunks.

- Personal Lines provide dedicated outside lines for multiline voice terminal users. Personal lines are accessed via a dedicated feature button. Up to 16 terminals may share a personal line.

- Third-Party Call Setup allows PCs to set up calls between a System 25 voice/data terminal and any other facility. A PC application program could use this capability to retrieve information from a data base.

Last Number Dialed, Repertory Dialing, and Speed Dialing are also applicable to dialing and managing internal calls. Personal lines provide both incoming and outgoing service.

Internal Call Movement

Typically, about 40 percent of PBX calls are internal calls, call transfers to another location, conference of multiple locations, temporarily suspended calls, page to locate the called party, etc. Rapid placement of internal calls and easy call movement from the answering station to a new station are supported with numerous features in System 25.

To provide easy internal call setup, System 25 provides the following features.

- Direct Station Selection (DSS) allows one-button access from a multiline voice terminal to another voice terminal, a pooled facility, paging zone, or DGC group. The DSS status LED reflects the idle/busy status of the associated termination point. This feature is used to track and contact frequently called associates.

- Automatic Intercom allows multiline voice terminal users to call each other by use of a dedicated line appearance. A private dedicated path ensures that a path is always available. This feature is frequently used in Boss/Secretary arrangements.

- The Dial Plan for System 25 is based on the concept that, whenever possible, calls should be placed to individuals rather than to pieces of equipment. To implement this concept, individuals are assigned PDCs and are allowed to sign in those PDCs at other terminals. The system automatically routes the call to the home terminal or signed-into terminal. This significantly increases the probability of reaching the called party. In addition, the Dial Plan is built on a flexible numbering scheme that allows the number of dialed digits to match assigned PDCs (2/3/4 digit dial plans) and to be administered to match telephone company assigned Direct Inward Dialing numbers.

Efficient internal call termination is supported with the following features.

- Distinctive Ringing provides various patterns of ringing to allow users to distinguish outside calls, inside calls, callbacks on queued calls, and calls set up at an associated data terminal.

- Hands-Free Answer on Intercom (HFAI) allows Speakerphone and HFAI terminals to auto-answer inside or attendant extended calls. With HFAI active, the set generates a tone burst over its speaker to alert the calling and called party of the call completion. Both parties may then converse; no action by the called party is required.

Frequently, the first termination point for a call is not its final destination. To support internal call movement, System 25 provides the following features.

- Bridging of System Access and Personal Lines allows calls to be passed in a manner that key system users are familiar with.

- Transfer allows a user to transfer any call to another voice terminal. This feature supports transfer of calls from the answering position to another location for completion of a transaction. Examples are secretary to boss, office to lab, broker to specialist, etc.

- Conference allows up to five parties (maximum two outside), including the originator, to participate in a call. This feature supports add-on of additional parties to a call for joint consultation, crisis management, schedule coordination, etc.

- Hold allows a user to suspend a call. The Hold feature allows users to temporarily disconnect from one conversation and either place or answer another call. Music or information bulletins may be provided to the held party. Called parties frequently use the hold period to access computer data bases, search categories, and/or consult with others via a second phone call.

- Following and Forwarding provide users with ways to answer their incoming calls while temporarily away from their home terminals.

- Park allows a user to place a call or conference on hold and then pick up the call from any voice terminal. The user can page another party to pick up the parked call or may move to another location and then re-access the call.

Data Communications

Small Business customers have started to integrate PCs into their day-to-day business operations. Businesses have found a need to access the data bases (sales, inventory, personnel) in these PCs from more than one location (both on- and off-premises). System 25 data features are specially engineered to enhance a user's ability to access data from multiple locations. System 25 has been designed to help these businesses use their personal computers, data terminals, and host computers more effectively by providing the

following features.

- Circuit switched data communications up to 19,200 bps (RS-232 interface) provide circuit switched connections from asynchronous data terminals, PCs, or host computers to host computers or network facilities. Users can be located and/or moved to any on-premises office equipped with the standard AT&T 4-pair wiring plan. Thus an asynchronous terminal or PC can have access to multiple host computers, remote data bases via a modem pool, and a local area network (STARLAN) via System 25's STARLAN NETWORK gateway.

- Packet switched data connections at 1 million bps over AT&T's STARLAN NETWORK local area network provide data transfer between client PCs and servers (PCs/host computers/printers, etc.) on the local area network (LAN). LAN users can be located and/or moved to any on-premises office equipped with standard AT&T 4-pair wiring. The LAN allows PCs to share facilities (printers, disk systems, modem pools, etc.).

- System 25's STARLAN NETWORK ACCESS software and STARLAN NETWORK gateway provide access to the STARLAN NETWORK for off-premises and occasional on-premises users. These users do not need to install a Network Access Unit (NAU) in their PCs to use the STARLAN NETWORK ACCESS software. The data transfer rate is limited to 9600 bps or, for off-premises users, by the modem.

  Note: System 25 is compatible only with Release 2 of the STARLAN NETWORK.

  LAN users can access hosts connected to System 25, or, if their System 25 is equipped with a modem pool, remote hosts. Finally, terminals and PCs connected to System 25 data ports can access host computers on the LAN.

Frequently a user needs to access a LAN data base at or from a remote location (home, motel, client office, branch, etc.). To support out-of-building access to computer data over network facilities or Off-Premise Station (OPS) lines, System 25 provides the following features.

- Modem pooling allows data terminals to communicate over analog facilities utilizing the standard dialing plan and provides full access to all network facilities, cost control mechanisms, ARS, and incoming call management tools (DID/attendant/DGC, etc.).

- Transfer to data allows a data call to be set up on a voice terminal and then to be transferred to a data terminal or computer, This feature can also be used to enter an account code for the data call.

- The System 25 STARLAN NETWORK gateway allows the LAN environment to be extended to occasional users or remote locations. Off-premises users can access the LAN utilizing all the network features, cost control mechanisms, and incoming call management facilities of System 25. The data transfer rate is governed by the modem.

Setting up data communications with PCs, host computers, and/or remote access can be a source of confusion for occasional users. The following special data features are provided by System 25 to assist the user in utilizing its rich set of data communications capabilities.

- The integrated voice-data dialing plan recognizes the different types of data endpoints (digital/ analog and remote/Iocal) in a connection and automatically inserts the required data communication equipment. In addition, autobauding supports the alignment of equipment with the capacity to transmit at different data rates.

- Station Hunting supports the use of a single dial code to access a group of host computer ports.

- Terminal Dialing provides the user with fast access to data communications via keyboard dialing at a terminal or PC.

- Command Mode provides a menu of data services supporting terminal dialing and display and control of user data port options. A user friendly Change Options menu is provided for user administration of data options.

- Expert Mode is an enhancement that provides an alternative method of accessing Command Mode functions. It eliminates the display of menus and allows multiple commands to be entered on a single line. Expert mode is suitable for use with computer-driven scripts for call setup.

- Communication Access Manager (CAM) is an MS-DOS* software application that provides a phone manager for placing voice and data calls for the user and VT100† terminal emulation. CAM may be used on either STARLAN NETWORK client workstations or on PCs connected to System 25. CAM has a 200-entry directory with one-touch dialing for both voice and data calls and auto-login capability for data calls to host computers. CAM's Remote Access feature provides password protected unattended access to PC files and electronic mail. File transfer is supported with the popular XMODEM protocol.

- STARLAN NETWORK ACCESS is an MS-DOS software application that allows PCs not connected to the STARLAN NETWORK to call through the System 25 STARLAN NETWORK Interface and run STARLAN NETWORK client software to access file and printer servers on the STARLAN NETWORK. ACCESS uses a PC's serial communications port to communicate with the STAR LAN NETWORK Interface. ACCESS is compatible with NETBIOS, permitting execution of most applications written for the IBM‡ PC Network and IBM Token Ring Network.

* Registered trademark of Microsoft Corp.

† Trademark of Digital Equipment Corp.

‡ Trademark of International Business Machines Corp.

Growth & Rearrangement

Historical data indicates that clients in the System 25 station range have a need for communications systems capable of significant growth and rearrangement. Clients need flexibility over the life of the system to easily add capacity, move stations, modify cost control options, etc. The architecture of System 25 was implemented with the objective to meet this need.

- Advanced Administration (optional) is an easy-to-use, menu driven personal computer software package for configuring the rich set of system options. Versions of this software are available for both MS-DOS and UNIX personal computers.

- Uniform Wiring Plan (four-pair) allows a building to be prewired for the rich set of AT&T Small Business PBX service offerings. This modular wiring plan supports client reconfiguration of an office with variations in station type (Analog, MET, MERLIN Communications System, futures) and data configurations (LAN, asynchronous, synchronous). It supports simultaneous voice and data from standard 4-pair modular jacks.

- System 25/75/85 Standard Architecture supports efficient growth with modular cabinets, universal carrier slots, nonblocking network, and uniform wiring plan. Every circuit slot in the system can be used for trunk cards or voice/data station cards. All these attributes allow the client to add future capability without breakage and re-engineering of existing equipment. Thus, the client is able to minimize initial investment while not restricting future growth.

Over time, the type of tools and facilities that a business utilizes changes. It is important that a PBX provide support for the full set of telephone company network options over its installed life, even when only a subset is initially used. Trunks link two switching systems, such as System 25 and the local Central Office or System 25 and another PBX. System 25 supports five different telephone company trunk interfaces to provide desired connectivity at minimum expense. Thus the opportunity exists to select the best trunk types, depending on tariffs and customer needs. For example:

- Loop Start (LS) trunks for public network access at minimum tariff. These trunks handle outgoing and incoming attendant calls, incoming DGC calls, outgoing pooled facility calls, and personal line calls.

- Ground Start (GS) trunks for public network access. These trunks handle the same type of calls as LS trunks. They provide protection against call reorigination without toll restriction, more reliable automatic route selection, virtual facilities, SMDR, and CAS. Simultaneous incoming and outgoing call seizure of the same trunk under heavy traffic conditions is essentially eliminated with ground start trunks. GS trunks should usually be selected in preference to LS trunks unless tariff considerations are overriding. Note, however, that Centrex Service requires LS trunks.

- Direct Inward Dialing (DID) trunks for dialing a station directly from outside (attendant assistance not required). Outside dial access to stations, trunks (optional), and answering groups (Direct Group calling) is provided.

- Tie Trunks for linking PBXs with dedicated private circuits for high volume calling. Dial access to stations, other trunks, answering groups (Direct Group Calling), and an Electronic Tandem Network endpoint capability are provided.

- Off-Premises Stations (OPS) allow a single-line voice terminal to be located remotely and connected to System 25 via arrangements with the local telephone company. This service is used to provide users at secondary sites (or their residences) many of the same features as an on-premises single-line station.

To enhance the usage and control of the above set of network facilities, System 25 provides the rich set of access features outlined in the Outgoing Business Communications section. In addition, System 25 can support networking between systems by:

- Serving as an endpoint on an electronic tandem network (ETN) using its tie trunks and flexible dialing plan.

- Serving as an off-network or on-network access point with its dial access/transfer between tie trunks and telephone company trunks (LS/GS/DID). This allows usage of tie trunks to reach a distant System 25 and then connect through that System 25 to local telephone company facilities to complete the call.

To support efficient utilization of trunks, they can be grouped together (up to 16 groups) if all trunks in the group perform the same function. This resource pooling provides better service with a given number of trunks, and simplifies administration and calling.

Types of trunks that can be assigned in System 25 are as follows.

- Central Office, which provide a link with the local telephone company for incoming and outgoing calls (LS/GS)

- Foreign Exchange (FX), which connect to a CO other than the local CO for high volume calling to/from a distant location

- Wide-Area Telecommunications Service (WATS), which connect to an Outward WATS office or a dial “800” Service Office

- Direct Inward Dialing (DID), which provide incoming service from a CO to directly access a station or facility (STARLAN NETWORK interface, trunk group)

- Tie, which provide a link with another private switching system.

To support efficient utilization of this rich set of network options, System 25 provides the functions outlined in the Incoming and Outgoing Business Communications sections.

Conclusions

System 25 has been targeted at providing excellent small business communications capability at the right price. The thousands of systems in service in the first 2 years of production have confirmed that these capabilities are an excellent match with small business customers' communications needs.

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FEATURES AND SERVICES

Introduction

This section describes the System Features, Station Features, Network Features, Data Features, and Attendant Features of AT&T System 25. It also covers certain services that support and implement the features; included in this category are the digital tape unit, the dial plan, system administration, and system maintenance. A general discussion of data topics is also provided.

The feature descriptions are arranged in alphabetical order, regardless of the feature group to which they belong. Information for each feature is presented under one or more of the following five subheadings: Description, Considerations, Interactions, Administration Requirements, and Hardware Requirements. Headings that are not applicable are omitted.

- **Description**

  Defines the feature, describes what it does for the user, and how it is used.

- **Considerations**

  Discusses the applications and benefits of the feature, followed by feature parameters and factors to be considered when the feature is used.

- **Interactions**

  Lists and briefly describes other features that can affect the feature being described. Interacting features are those that:

  — Depend on each other—One of the features must be provided if the other one is.

  — Cannot coexist—One of the features cannot be provided if the other one is.

  — Affect each other—The operation of one feature modifies, or is modified by, the operation of the other.

  — Enhance each other—The features, in combination, provide improved service to the user.

- **Administration Requirements**

  States whether or not administration is required and lists items requiring administration.

- **Hardware Requirements**

  Lists any additional hardware needed to use the feature.

Symbols Used in Illustrations

Many feature descriptions in this section contain illustrations of equipment and connections. In the connection figures, modular jacks are shown as triangles; 25-pair cable connectors are indicated by shaded blocks. Unterminated wiring that requires cutdown or other termination does not have symbol designations. The 103A Connecting Block is a typical modular wall jack that provides cutdown connections for building (station) wiring.

Feature Tables

Tables 2-A through 2-E list all the features of System 25. Each feature is specified as Standard or Optional.

Standard features are built into the system. They are always provided but may require administration to make them operational. Standard features are identified in the feature tables by the letter S.

Optional features require both administration and additional equipment. Music-On-Hold is an example. Optional features are identified by the letter O.

Bracketed words in the tables are the standard labels of the associated feature buttons. These labels are also used in the feature descriptions.

System Features

System features (Table 2-A) are those that affect the entire operation of the system.

Table 2-A. System Features

FEATURE NAME	FEATURE TYPE
Call Accounting System (CAS)	O
Call Management System (CMS)	O
Dial Plan	S
Dlctatlon System Access	O
Digital Tape Unit	O
Direct Group Calling	S
Direct Group Calling Delay Announcement	O
End-to-End Signaling	S
Extended Stations	O
External Alerts	O
Integrated Solution	O
Intercept Treatment With Reorder Tone	S
Interdigit Timeouts	S
Music-On-Hold	O
Night Service (Directed and TAAS)	S/O*
Night Service Delay Announcements	O
Out-Of-Building Stations	O
Paging System Access	O
Personal Dial Codes	S
Pooled Facility-Dial Access	S
Power Failure Transfer	O
Remote Administration Interface	O
Remote Initialization and Maintenance Services (RIMS)	S
Station Message Detail Recording (SMDR)	O
System Administration	O
System Maintenance	S
Touch-Tone and Dial Pulse Service	S
deice Message System	O

* S/O - Standard for Directed, Optional for TAAS Night Service.

Station Features

The many Station Features (Table 2-B) available allow individual user needs to be met. As these needs change, assigned features can also be changed. Station Features provide many important services that help save time and make calling more convenient.

Table 2-B. Station Features

FEATURE NAME	SINGLE-LINE TERMINAL	MULTILINE TERMINAL BUTTON LABEL	FEATURE TYPE
Account Code Entry, Forced (FACE)	X		S
Account Code Entry, Optional	x	[ACCT ENTRY]	S
Automatic Intercom		[AUTO ICOM]	S
Bridging of System Access Buttons		x	S
Busy-to-Idle Reminder		x	S
Callback Queuing	X	X	S
Calling Restrictions	x	x	S
Call Accountability	x	x	S
Call Progress Tones	x	x	S
Call Waiting	X		S
Conference	x	[CONFERENCE]	S
Conference Drop	X	[DROP]	S
Coverage-Group	x	[COVER-GRP]	S
Coverage-Individual	x	[COVER-IND]	S
Dial Access to Message Waiting Indications	x	x	S
Direct Station Selection (DSS)		[DSS] or [FLEX DSS]	S
Directory		[DIRECTORY]	O
Display		[SCROLL]	O
Distinctive Ringing	x	x	S
Exclusion		[EXCLUSION]	S
Following	x	x	S
Forwarding	x		S
Hands-Free-Answer On Intercom (HFAI)		[AUTO ANS]	O
Headset Adapter Adjunct	x		O
Hold	x	[HOLD]	S
Inspection		[INSPECT]	O
Last Number Dialed		[LAST # DIALED]	S
Leave Word Calling (LWC)	x	[LEAVE WORD]	O
Line Selection		x	S
Line Status And I-Use Indications			S
Local Display		[LOCAL]	O
Manual Signaling		[SIGNAL]	S
Messaging Services	x	x	S
Park	x		S
Personal Lines		[PERS LINE]	S
Pickup	X		S
Pooled Facility-Button Access		[FACILITY]	S

Table 2-B. Station Features (Contd)

FEATURE NAME	SINGLE-LINE TERMINAL	MULTILINE TERMINAL BUTTON LABEL	FEATURE TYPE
Program	x	x	S
Recall	x		S
Repertory Dialing		[REP DIAL]	S
Send All Calls		[SEND ALL CALLS]	S
Speaker (Spokesman Service)		[SPEAKER]	S
Speakerphone Adjunct	x	x	O
Speed Dialing	x	x	S
Station Hunting	x		S
Station-To-Station Message Waiting		[MSG WAIT]	S
Test			S
Transfer	X	[TRANSFER]	S
Trunk-To-Trunk Transfer	x	x	S

Network Features

This group of features (Table 2-C) supports communications with the public network and with other locations in the private network of which System 25 can be a part.

Table 2-C. Network Features

FEATURE NAME	FEATURE TYPE
Automatic Route Selection	S
Direct Inward Dialing	O
Off-Premises Stations	O
Remote Access	S
Tandem Trunking	O
Tie Trunks	O
Trunk Groups	S
Virtual Facilities	S

Data Features

Data Features (Table 2-D) support the switched data services of the system. Data services provide switched connections between analog and digital data endpoints.

Table 2-D. Data Features

FEATURE NAME	MULTILINE TERMINAL BUTTON LABEL	FEATURE TYPE
Command Mode		S
Communications Access Manager		O
Data Call Setup		S
Data Terminal Dialing		S
Expert Mode		S
Modem Pooling		O
AT&T STARLAN NETWORK Access		O
Third-Party Call Setup		S
Transfer to Data	[DATA]	S
User Changeable Options		S

Attendant Features

Attendant Features (Table 2-E) are available to the attendant using the Direct Trunk Attendant Console (DTAC) or the Switched Loop Attendant Console (SLAC) and the optional Direct Extension Selector Console. In addition, most muitiline voice terminal station features are available to the attendant.

Table 2-E. Attendant Features

FEATURE NAME	CONSOLE BUTTON LABEL	FEATURE TYPE
Attendant Call Extending	[START]	S
Attendant Camp-On		S
Attendant Cancel	[CANCEL]	S
Attendant Console, Direct Trunk		O
Attendant Console, Switched Loop		O
Attendant Direct Extension Selection		O
Attendant Forced Release (SLAC only)	[FORCED RELEASE]	S
Attendant Join (SLAC only)	[JOIN]	S
Attendant Message Waiting (DTAC)	[ATT MSG]	S
Attendant Message Waiting (SLAC)	[ATTENDANT MESSAGE WAITING]	S
Attendant Position Busy	[POS BUSY]	S
Attendant Release	[RELEASE]	S
Attendant Return-Coverage-on-Busy	[RTN-BUSY]*	S
Attendant Return-Coverage-on-Don't-Answer	[RTN-DA]*	S
Attendant Source/Destination (SLAC only)	[SOURCE], [DEST]	S
Attendant Splitting One-Way Automatic		S
Attendant System Alarm Indication	[ALARM]	S
Message Center-Like Operation (SLAC only)		S
Night Service	[NIGHT]	S

* This button is assigned on the DTAC only,

Account Code Entry, Forced

Description

This feature forces selected station users to enter account codes before dialing certain calls out of System 25. Users at stations that have Forced Account Code Entry (FACE) are required to enter account codes either for all outgoing calls or for just “dial 0 or 1” toll calls. The code entries appear in the ACCOUNT field of the SMDR records.

To make a FACE-restricted call, the user must dial the Account Code Entry access code *0 followed by an account code *before* dialing the rest of the call. The account code entry is terminated when the number of digits entered equals the number administered for system account codes or when # is entered. The user hears second dial tone after the code is entered and can then dial the necessary access codes and other numbers to reach the destination.

If the user makes an error while entering the account code, the procedure can be corrected by dialing *0 followed by the correct account code.

The user receives reorder tone when an account code is required on a call but not entered.

Considerations

FACE ensures that specified outgoing calls include information (project, client, department, etc.) to be used for accounting and billing purposes.

The voice terminal user cannot use the Account Code Entry feature button for forced entry. This button is used with the Optional Account Code Entry feature only.

An account code entry cannot be forced for the following types of calls:

- Personal Line calls
- Direct Facility Access calls
- Remote Access
- Calls to 911 and the three ARS-administered emergency numbers, when using ARS.

FACE requirements apply to calls using these facilities and features:

- Repertory Dialing
- Personal/System Speed Dialing
- ARS (nonemergency) and pooled facility access codes
- Trunk calls using Conference or Transfer.

The system does not check the validity of account codes. It checks only for the proper number of digits or the code terminator #.

Calls that do not require FACE can still be assigned an account code, as in previous releases of System 25. Refer to the "Account Code Entry, Optional" feature description in this manual for the procedures.

Interactions

The following features interact with Forced Account Code Entry.

Bridging of System Access Buttons: Calls made from Bridged Access (BA) buttons on a bridging station follow the FACE restrictions of the bridging station, not of the associated principal station.

Call Accountability: The account code entry may be made before or after the Call Accountability entry. Dial tone is returned to the user after either entry.

Callback Queuing: An account code entered before queuing is saved for SMDR.

Conference: Calls can be conference in both directions between a FACE-restricted station and a non-FACE station.

Display: When a user activates the Forced Account Code Entry feature by dialing *0, the system displays the prompt ACCT?. As the user enters the account code, the digits are displayed to the right of the prompt. If the number of digits exceeds 9, the system automatically scrolls to Screen 2; the continuation character "-" and the remaining digits appear on Screen 2.

The prompt and digits remain displayed until one of the following occurs:

— The user enters either "#" or the administered number of code digits.

— The user restarts the Account Code Entry feature by dialing *0 to correct an erroneous entry.

— The system time-out for Account Code Entry is reached.

— The user selects another button that overwrites the display.

Forwarding: Stations with FACE administered for all calls cannot forward calls to any outside numbers, Stations with FACE administered for "dial 0 or 1" calls can forward calls to any outside number except for "dial 0 or 1" numbers.

Intercept Treatment with Reorder Tone: The user receives reorder tone when an account code is required on a call but is not entered.

Last Number Dialed: The access code *0 and the account code are not stored by this feature.

Remote Access: Remote access callers cannot enter account codes.

Third-Party Call Setup: If the source station is FACE-restricted, the third-party data terminal must prefix the outside destination number with *0 and an account code.

Transfer: Calls can be transferred in both directions between a FACE-restricted station and a non-FACE station.

Administration Requirements

Account code entry is administered on a per-station basis—Optional, Forced for all Outgoing Calls, or Forced for Dial 0 or 1 Toll Calls Only; default = Optional.

FACE cannot be administered for data ports.

Account Code Entry, Optional

Description

Optional Account Code Entry allows voice terminal users to associate an account code with incoming and outgoing calls. The account code is appended to the SMDR call record and can be used later for accounting or billing purposes.

For an incoming call, the user must enter the account code at the end of the call. For an outgoing call, the user has a choice of entering the code at the beginning of the call, before the destination is dialed, or at the end of the call. An account code entry is terminated when the number of digits entered equals the number administered for system account codes, when # is entered, or when the user hangs up. The procedures for associating an account code with a call are as follows:

- *Single-Line Voice Terminal User*

  Get dial tone (by going off-hook at the beginning of a call or by flashing the switchhook before hanging up) and dial *0; then dial the account code directly or dial a System or Personal Speed Dialing Number that contains the account code. If the code is dialed incorrectly (before the last digit), redial *0 and the correct number.

- *Multiline Voice Terminal User*

  At the beginning of an outgoing call, get dial tone and dial *0; then dial the account code directly or dial a System or Personal Speed Dialing Number that contains the account code. If the code is dialed incorrectly (before the last digit), redial *0 and the correct number. At the end of a call, press ACCT ENTRY and enter the code before hanging up. A Repertory Dialing (REP DIAL) button can also be used to enter an account code. If the code is dialed incorrectly (before the last digit), press ACCT ENTRY again and dial the correct number.

When the correct number of account code digits has been entered (or # is entered to signal end-of-dialing), confirmation tone is returned to the user, and the account code is appended to the SMDR call record.

Considerations

Optional Account Code Entry provides an easy method of allocating the costs of specific calls (and associated staff time) to the correct project, department or user. The account code is appended to the SMDR call record and sent to the SMDR output channel.

Account Codes can contain up to 15 digits.

The system does not check the validity of account codes. It only checks for the proper number of digits or the code terminator #.

If the user is active on a call, invoking the feature will drop the call.

Erroneous account codes that are not corrected before the last digit is entered are recorded and cannot be changed. Partial account codes entered by going on-hook before completing the entry are recorded and cannot be changed.

If, before all digits have been entered, (1) the user goes on-hook, (2) a button other than ACCT ENTRY is pressed, or (3) 30 seconds have elapsed since the feature was invoked, the SMDR call record will show the digits dialed up to that point.

Optional Account Code Entry cannot be invoked for a call on hold.

Interactions

The following features interact with Optional Account Code Entry.

Bridging of System Access Buttons: Account codes can be entered for incoming or outgoing calls on Bridged Access buttons using normal feature operations.

Callback Queuing: An account code entered before queuing is saved for SMDR.

Conference: If more than one user attempts to enter an account code on a Conference Call, the first to enter a code will prevail.

Display: When a user activates the Account Code Entry feature by dialing *0 or pressing ACCT ENTRY, the system displays the prompt ACCT?. As the user enters the account code, the digits are displayed to the right of the prompt. If the number of digits exceeds 9, the system automatically scrolls to Screen 2; the continuation character "-" and the remaining digits appear on Screen 2.

The prompt and digits remain displayed until one of the following occurs:

- The user enters either "#" or the administered number of code digits.
- The user restarts the Account Code Entry feature by dialing *0 or pressing ACCT ENTRY again, to correct an erroneous entry.
- The system time-out for Account Code Entry is reached.
- The user selects another button that overwrites the display.

Remote Access: Remote access callers cannot enter account codes.

Repertory Dialing: An account code can be stored on a REP DIAL button.

Speed Dialing: An account code can be stored in a System or Personal Speed Dialing code.

Transfer: A user can transfer a call to another user, then, before hanging up, enter an account code. Subsequent account code entries for the same call will be ignored, even though confirmation tone has been returned.

Administration Requirements

System:

- Assign number of Account Code digits (0-15; default = 15).

Voice Terminal Port:

- Multiline terminals—Assign Account Code Entry button.
- Single-line terminals—no administration required.

Hardware Requirements

Requires an RS-232 compatible 80-column ASCII (serial) printer or other device to output Station Message Detail Recording (SMDR)/Account Code entries.

Attendant Call Extending

Description

This feature allows the attendant to put a call in a special hold condition, call another station, then connect the two calls together. The attendant can withdraw from the connection and separate the call from the console or remain connected to the other parties. Attendant Call Extending is a feature used at either a Direct Trunk Attendant Console (DTAC) or a Switched Loop Attendant Console (SLAC).

Note: In general, the attendant *should not* use the TRANSFER button, which invokes the standard multiline voice terminal Transfer feature, to extend calls. If Transfer is used, busy or unanswered calls cannot return to the attendant console for further handling.

The attendant, after placing or answering a call, can use Procedure 1 or 2 to extend this call to an inside extension or Procedure 1 to extend it to an outside number:

1. Press START to place the incoming call on hold via the Attendant Splitting One-Way Automatic feature. After receiving Dial Tone, the attendant then dials the requested inside or outside number.

   or

2. Press the Selector Console Group Select and Direct Extension Selection (DXS) buttons associated with the requested inside station. This operation is equivalent to pressing START and dialing the extension.

If ringing tone is heard, the attendant presses RELEASE (Manual Release) to connect the caller to the ringing line and separate the call from the console. As an alternative, a DTAC attendant or a SLAC attendant (with Automatic Release administered) can go straight to another call by pressing any facility button, such as System Access, Loop, Automatic Intercom, or an outside line; this completes the call extending procedure. (if a SLAC attendant has Automatic Hold administered instead of Automatic Release, pressing a facility button simply puts the incoming call on hold and does not extend it.)

The attendant has the option of staying connected to the ringing line to announce the call before connecting the two parties. The attendant can then release or (SLAC only) join the other parties in a 3-way connection by using the Attendant Join feature.

If busy tone is heard and Attendant Camp-On (see associated feature description) is not desired, the attendant presses CANCEL and is reconnected to the calling party.

If busy tone is heard on a call to an inside station and Attendant Camp-On is desired, the attendant presses RELEASE. The called party hears a tone burst, and the call waits at the called voice terminal. When a busy single-line station goes on-hook, or a busy multiline station System Access button becomes idle, the call automatically begins ringing at the station.

Calls extended to an *idle* voice terminal that are not answered within a specified time return to the Attendant Console on an idle LOOP button (SLAC only) or on the Return-On-Don't-Answer (RTN-DA) button (DTAC only). Calls camped-on at a *busy* voice terminal that are not answered within a specified time return to the Attendant Console on an idle LOOP button (SLAC only) or on the Return-On-Busy (RTN-BUSY) button (DTAC only). If a SLAC is not available to incoming calls (busy on another call, in Position Busy mode, etc.), a returning call remains in the console queue until the console can handle it. If the Return buttons on a DTAC are busy, an extended call remains at the called terminal until that button becomes idle.

Considerations

Attendant Call Extending allows the attendant to utilize the additional attendant related features such as Attendant Splitting One-Way, Release, Cancel, Return-On-Don't-Answer, Return-On-Busy, Forced Release (SLAC), Join (SLAC), and Source/Destination (SLAC).

For information on related Attendant Features (Table 2-E), refer to the individual feature descriptions.

Interactions

The following feature interacts with Attendant Call Extending.

Forwarding: Calls extended by an attendant to a forwarding station will be given normal Forwarding treatment.

Administration Requirements

System:

- Number of seconds before a Camped-On call returns to the Attendant Console (1-120 or 0 for no Attendant Camp-On; default = 30).
- Number of rings before unanswered call returns to the Attendant Console (1-31; default = 5).

Attendant Camp-On

Description

This feature allows the attendant to extend a trunk call to a busy voice terminal and leave it waiting or "camped on" there. After hearing busy tone, the attendant presses RELEASE to camp-on this call at the busy terminal. When this is done, a burst of tone is heard in the handset of the called terminal and the caller is placed on hold (hearing music-on-hold if available). When a System Access button at a multiline set becomes idle or a single-line terminal hangs up, the camped-on call is connected automatically and ringing begins.

Considerations

A camped-on call can be answered by a busy single-line user without losing the current call by momentarily pressing the switchhook (which places the current call on hold) and then dialing *9. Multiline terminal users cannot do this. However, if they have a System Access-Originate Only button, they can place all other calls on hold, go off-hook on that button and dial *9 to pick up the camped-on call.

If the camped-on call is not answered within a specified time, the call will be returned to the Attendant Console in one of the following ways:

- *Switched Loop Attendant Console*

  The call returns to the common queue, where it remains until the console can receive it at a LOOP button.

- *Direct Trunk Attendant Console*

  The call returns to the Return-On-Busy (RTN-BUSY) button. If that button is busy, the call remains camped-on at the called terminal until the RTN-BUSY button of the console becomes idle.

For information on related Attendant Features (Table 2-E), refer to the individual feature descriptions.

Interactions

The following features interact with Attendant Camp-On.

Call Waiting: Trunk calls camped onto a station by an attendant are given priority over other waiting calls.

Callback Queuing: Trunk calls camped onto a station by an attendant are given priority over queued calls.

Coverage: If the called party is a sender in a Coverage group and all receivers of the Coverage group are busy, the call will camp onto the originally-dialed station. Once camped-on, calls will no longer receive coverage even if a coverage receiver becomes idle.

Direct Group Calling: The attendant can camp-on more than one call per DGC group. Voice terminals in the group do not receive a burst of tone when a call is camped on. Trunk calls camped onto a busy DGC group go into the DGC queue and are eligible for delay announcement and music-on-hold.

Direct Inward Dialing (DID): DID calls can be covered by the attendant and then given Camp-On treatment. They do not automatically receive Call Waiting.

Station Hunting: If the called party is a member of a hunt group and all members of the group are busy, the call camps onto the originally-dialed station. Once camped-on, calls will no longer hunt even if another member of the hunt group becomes idle.

Administration Requirements

System:

- Number of seconds before a camped-on call returns to the Attendant Console (1-120 or 0 for no Attendant Camp-On; default = 30).

Attendant Cancel

Description

This feature allows the attendant to terminate an attempt to extend any incoming call if the called station does not answer, or if the station answers but declines to accept the call. The attendant presses CANCEL and is automatically reconnected to the calling party. The call can then be ended by hanging up or by pressing RELEASE.

Pressing CANCEL when the Start facility is not active will be ignored.

Considerations

Attendant Cancel allows the attendant to terminate a call transfer attempt and return to the incoming held party via a one-button operation. This enhances the attendant's ability to handle calls quickly and efficiently.

Attendant Console, Direct Trunk

Description

In System 25 the Attendant Console is used to answer incoming trunk calls that are not directed to specific user stations, to answer calls from inside users, to extend calls to inside stations and outside numbers, and to assist system users in placing outgoing calls and setting up conferences. The attendant can also manage and monitor some areas of system operation. System 25 supports either the Direct Trunk Attendant Console (DTAC) or the Switched Loop Attendant Console (SLAC), which is covered in the next feature description in this manual. Consoles of both types cannot be installed in the same system.

The DTAC (Figure 2-1) can be one of the Merlin Communications System multiline voice terminals listed below, administered with special features, buttons, and capabilities to serve as an attendant position.

- 7305H02D (34 programmable feature buttons each with I-use and status LEDs)
- 7305H03B (34 programmable feature buttons; built-in speakerphone)
- 7316H01A (34 programmable feature buttons each with I-use and status LEDs; built-in speakerphone)

In addition to the attendant features, most standard multiline terminal features are also available. (Refer to the Hardware Description section of this manual for a complete identification of the external controls, indicators, and components of the DTAC voice terminal.)

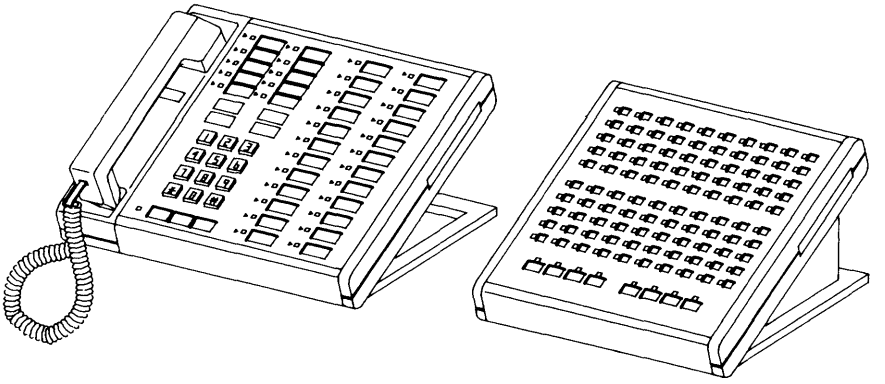


Figure 2-1. Typical Direct Trunk Attendant Console Position

The DTAC is always equipped with the following feature buttons that provide unique attendant console functions. Each button has a green status LED that indicates when the feature is activated.

- **Start [START]:** Initiates the call extending process by placing a caller on hold and providing internal dial tone to the attendant
- **Cancel [CANCEL]:** Terminates the "Start" operation and reconnects the attendant to the calling party.
- **Release [RELEASE]:** Releases the attendant from an active call and completes the call extending process.
- **Return-On-Busy [RTN-BUSY]:** Camped-on calls are returned to the console on this button if not answered within a specified interval.
- **Return-On-Don't-Answer [RTN-DA]:** Extended calls not answered are returned to the console on this button if not answered within a specified interval.
- **Attendant Message Waiting [ATT MSG]:** Used by the attendant to remotely control Message LEDs on voice terminals.
- **Alarm [ALARM]:** The associated status LED flashes when a system trouble has been detected; the LED can be changed from flashing to steadily lit by pressing the button.

Two other attendant-only features are assigned to console feature buttons if required, Position Busy [POS BUSY] and Night Service [NIGHT]. In a dual attendant console system, Position Busy removes an Attendant Console from service. Only one of two consoles can be in the "Position Busy" mode at a time. When Night Service is activated, attendant-seeking calls can ring a night bell, can be directed to assigned voice terminals, or can be sent to a night service announcement.

Dual Console Operation

A System 25 can be equipped with up to two DTACs that operate simultaneously when both are in service. If the system has two attendant consoles, one is called the first attendant console; the other is called the second attendant console. The calls in the following list will be routed to the first attendant console.

- Dial "0" calls
- DID calls to unassigned numbers (when administered to route to the attendant)
- Calls to Floating Personal Dial Codes (FPDCs) not signed in anywhere (if administered to route to the attendant)

If the first attendant has activated the Position Busy feature or is busy on all System Access buttons, these calls will be routed to the second console. If that console is also busy on all System Access buttons, busy tone is provided to the calling party.

System users and DID callers can reach a particular attendant by dialing the Personal Dial Code (PDC) assigned to the desired attendant.

Position Busy

A POS BUSY button can be assigned to each console; this permits selection of one of two modes of operation: (1) simultaneous operation or (2) only one Attendant Console active. (Note that only one console is allowed to be inactive at any given time.) An associated POS BUSY status LED is lighted when the console is inactive. Ringing is disabled on all trunk terminations on the rightmost two columns of buttons of the inactive console. Ringing disabled on an inactive console will be enabled on the active console for those trunks with dual appearances (appearances on both consoles). All other features on all buttons, including those on the associated Attendant Direct Extension Selector Console will continue to function normally even though the console is inactive.

Considerations

Direct trunk operation means that trunks are terminated on individual buttons, called Personal Line buttons, where outside calls are answered and originated. The console can have several incoming calls ringing simultaneously.

Each console can also have an optional Attendant Direct Extension Selector Console to enhance internal calling. The Selector Console is covered in the “Attendant Direct Extension Selection” feature description.

For information on related Attendant Features (Table 2-E), refer to the individual feature descriptions.

Interactions

The following features interact with Direct Trunk Attendant Console.

Attendant Console, Switched Loop: A DTAC cannot operate in the same system with a Switched Loop Attendant Console.

Bridging of System Access Buttons: A DTAC cannot serve as a principal station.

Callback Queuing: The attendant can queue calls that are extended using the normal START-RELEASE button operation. However, calls originated using only the START button (no other call put on hold) cannot be queued.

Call Waiting: Calls cannot wait at a DTAC.

Display: The DTAC does not support Display.

Administration Requirements

System:

- Display attendant position number {first or second).
- Assign number of rings before unanswered calls return to the Attendant Position (1-31; default = 5).
- Force DID calls to unassigned numbers to ring at the Attendant Position (yes or no; default = yes).
- Force calls to FPDCs that are not signed in anywhere to ring at the Attendant Position (yes or no; default = yes).
- Assign number of seconds before an unanswered Camped-On Call returns to the Attendant Console (1-120 or 0 for no Attendant Camp-On; default = 30).

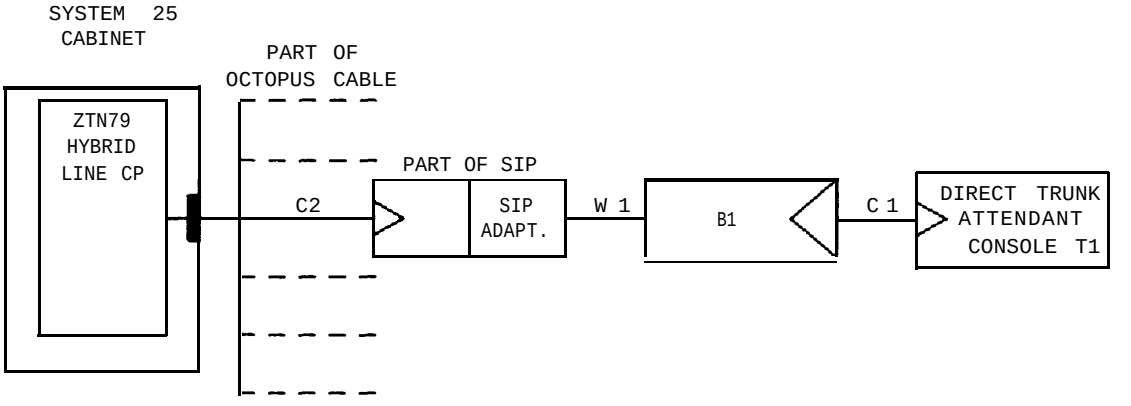
Attendant Console (Voice Terminal) Port:

- Assign voice terminal type (309).
- Assign buttons for Night Service and Position Busy, if required. Attendant Message Waiting is defaulted to button 14, but can be assigned to any programmable button.
- The following buttons are predefine on the Attendant Console and are not administrable: ALARM, RTN-DA, RTN-BUSY, START, CANCEL, and RELEASE.
- Trunk terminations; the following is required for each trunk terminated on the console (administered as Personal Line appearances; DID trunks cannot be terminated on a DTAC):
  - Trunk Number
  - Make this the Principal Station (owner) of the trunk (yes or no).
  - Enable Ring (yes or no).

Hardware Requirements

Each console requires a port on a ZTN79 ATL Line circuit pack.

Figure 2-2 provides a connection diagram for the DTAC.



LEGEND:

B1 — TYPICAL-103A CONNECTING BLOCK*
C1 — MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 — OCTOPUS CABLE (WP90780) - PEC 2720-05P
T1 — 34-BUTTON VOICE TERMINAL: 7305H02D - PEC 3162-417,
7305H03B - PEC 3162-BIS, OR 7316H01A - PEC 3167-34B
W1 — 4-PAIR INSIDE WIRING CABLE*
* — FURNISHED BY INSTALLER

RANGE: WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED >1000 FEET)

Figure 2-2. Direct Trunk Attendant Console Connections

Attendant Console, Switched Loop

Description

In System 25, the Attendant Console is used to answer incoming trunk calls that are specified to ring at an attendant position, to answer calls from inside users, to extend calls to inside stations and outside numbers, to assist system users in placing outgoing calls, and to set up conferences. The attendant can also manage and monitor some areas of system operation. System 25 supports the Switched Loop Attendant Console (SLAC) or the Direct Trunk Attendant Console (DTAC), which is covered in the preceding feature description of this manual. Consoles of both types cannot be installed in the same system.

The SLAC (Figure 2-3) can be one of the Merlin System multiline voice terminals listed below, administered with special buttons, features, and capabilities to serve as an attendant console. In addition to the attendant features, most standard multiline terminal features are also available. (Refer to the "Hardware Description" section of this manual for a complete identification of all the external controls, indicators, and components of the SLAC voice terminals.)

- 7305H04C (34 programmable feature buttons; built-in speakerphone and display)
- 7317H01A (BIS-34D—34 programmable feature buttons each with I-use and status LEDs; built-in speakerphone and display)

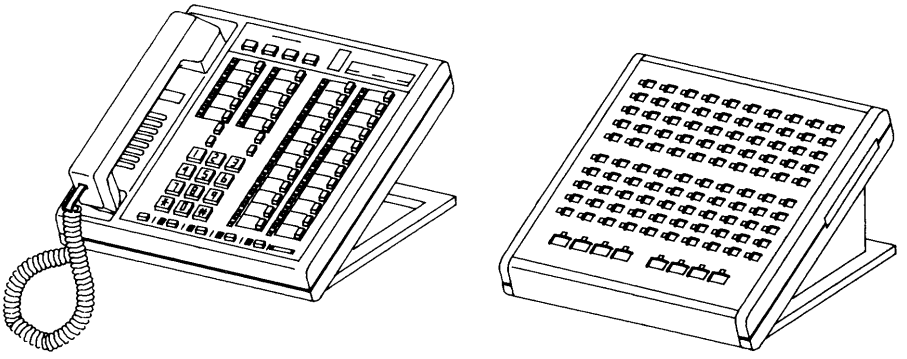


Figure 2-3. Typical Switched Loop Attendant Console Position

Associated with the SLAC are message center-like capability and display support. The message center feature provides for efficient handling of calls that should be sent to message takers. These calls are directed to a message center console position through administration of call type translations. Display service allows identifiers (names) to be assigned to extension numbers and trunks. The system then displays the appropriate information to the attendant when calls are processed at the console.

The Switched Loop Console derives its name from the ability of the system to hold incoming attendant-bound calls in a queue and switch them on voice loops to an available console. Calls are directed to a console in a preadministered, prioritized sequence. The SLAC differs from the DTAC in the following basic respects:

- It receives calls one at a time, regardless of the number of incoming calls to the system (at the DTAC, many incoming calls can be ringing simultaneously).
- It displays pertinent information about incoming and outgoing calls.
- It can serve as an attendant console, a message center, or a combination of both.
- It has speakerphone and Hands-Free Answer on Intercom (HFAI) capabilities.

Fixed Buttons (Figure 2-4)

The SLAC has five fixed line appearance, or "LOOP," buttons where all incoming calls are answered. Each button has a red I-use LED and a green status LED. These buttons represent voice links (loops) between the console and the switch. The loops also provide the paths for outgoing calls.

In addition to the LOOP buttons and standard multiline terminal buttons (HOLD, TRANSFER, etc.), the console is equipped with the following feature buttons that provide unique attendant functions. On the deluxe SLAC, all of these buttons have both I-use and status LEDs; on the basic SLAC, only the buttons specifically noted have LEDs.

- **Start [START]:** Initiates the call extending process by placing a caller on hold (on the Source button) and providing internal dial tone to the attendant.
- **Cancel [CANCEL]:** Terminates the "Start" operation and reconnects the attendant to the calling party (on the Source button).
- **Release [RELEASE]:** Releases the attendant from an active call and completes the call extending process.
- **Source [SOURCE]:** Reconnects the attendant to the calling party after a call has been initiated to the called party but before the two parties have been connected together. (i-use and status LEDs on basic SLAC.)
- **Destination [DEST]:** Connects the attendant to the called party again after the attendant has operated the Source button to speak to the calling party. (I-use and status LEDs on basic SLAC.)
- **Join [JOIN]:** Joins together (in a 3-way connection) the attendant and the other parties in an extended call.
- **Forced Release [FORCED RELEASE]:** Drops all active parties from a call.
- **Last Number Dialed [LAST # DIALED]:** Redials the last number dialed.

- **Position Busy [POS BUSY]:** Temporarily removes the attendant position from service. (I-use and status LEDs on basic SLAC.)
- **Attendant Message Waiting [ATTENDANT MESSAGE WAITING]:** Used by the attendant to remotely control Message LEDs on voice terminals. (Status LED only on basic SLAC.)
- **Alarm [ALARM]:** The associated status LED flashes when a system trouble has been detected; the LED can be changed from flashing to steadily lit by pressing the button. (I-use and status LEDs on basic SLAC.)
- **Inspect [INSPECT]:** Puts the display into a mode for inspecting the status or stored information of certain buttons. (Status LED only on basic SLAC.)
- **Scroll [SCROLL]:** Causes display to present additional call information, when available.
- **Local [LOCAL]:** Allows display to be used for clock and calendar functions.

The buttons not assigned to normal voice terminal functions or to attendant functions are defaulted to the Flex DSS feature. One of these programmable buttons can be assigned to Night Service, if the feature is required, and any of the others to multiline voice terminal features.

Programmable Feature Buttons (Figure 2-4)

The features in the following list can be assigned to the programmable feature buttons. On the deluxe SLAC, each of the programmable buttons is equipped with an I-use LED and a status LED. On the basic SLAC the buttons do not have LEDs.

Account Code Entry	Exclusion
Agent Status for CMS	Flex DSS
Auto Answer	Leave Word Calling
Auto Intercom	Manual Signaling
Call	Next
Direct Facility Access	Repertory Dialing
Direct Station Selection (DSS)	Station-to-Station Message Waiting
Directory	Transfer to Data

Flex DSS and Repertory Dialing can be programmed with dialable numbers by the attendant. When a call is placed using a Flex DSS button or a Repertory Dial button, one of the five switched loops is automatically selected for routing the call to the switch.

Display (Figure 2-4)

The SLAC contains an alphanumeric call information display. This module is built into the top of the console. It contains a 16-character 5x7 dot matrix liquid crystal display, timer controls, and a thumbwheel Contrast adjustment. Timer functions are available only when the attendant presses the Local button. The Time/Timer Exit button allows the user to select ordinary clock/calendar display or a timer. In the Time mode, Set, Fwd, and Rev are used to set the clock. In Timer mode, Start and Stop are used to time events.

The primary purpose of the console display is to provide the attendant with descriptive information about incoming and outgoing calls. This information includes extension numbers and associated names, trunk identifiers, reasons for call return and redirection, and number of calls waiting in the queue for service. Refer to the “Display” feature description for a detailed discussion of call information displays.

The console display also provides access to the system’s integrated directory and allows the attendant to search for the extension numbers assigned to users. Refer to the “Directory” description for information on this feature.

Switched Loop Operation

All calls that are intended for an attendant position are first routed by the system to a common queue where they wait to be sent to a console. In a configuration having two consoles, the same queue serves both consoles. When an attendant console becomes available to receive a call, the system removes a call from the queue and directs it to an idle loop on the console. Calls are selected from the queue on the basis of “first in/first out” and in accordance with administered priorities. An available attendant console is one that is not active on a call, has no calls ringing, has at least one LOOP button idle, is not in Position Busy or Inspect mode, and is not in a split condition.

In a two-console arrangement, each console can be administered to receive all types of calls or to receive only specific types. A call that can be received by either position goes to the first available attendant; when both are available, the call goes to the attendant who has been idle the longest time. If one console is in “Position Busy” mode, all calls (except Attendant PDC, Attendant PDC via DID, and DGC calls) direct to the other console.

An incoming call from the queue to a console appears on one of the five LOOP buttons; the attendant is alerted to the call by audible ringing, a steadily lighted red lamp, and a flashing green lamp. While the call is ringing and while the attendant is handling the call, the system will direct no more calls to the console. After the attendant ends or releases the call or puts it on hold, another call can come in on an idle button.

It should be emphasized that even when all LOOP buttons on a console are idle, only one call can be directed from the queue to the console. If the attendant puts a call on hold, that LOOP button is no longer available, but a new call can come in on another button that is idle.

Answering a Call on a LOOP Button

At an available SLAC, an incoming call will ring at an idle LOOP button automatically selected by the system. The attendant has only to lift the handset to answer the call; pressing the button is not necessary.

Placing a Call on a LOOP Button

In general, originating a call at an idle SLAC involves going off-hook and then dialing the desired number. If the console is not idle, the attendant can generally use one of these procedures:

- Split the active call (that is, put it on temporary hold by pressing the Start button) and place another call on the same button; this is the normal call-extending procedure.
- If the Automatic Hold feature is enabled, press another LOOP button to place a new call; the first call goes on hold.
- If the console does not have Automatic Hold (that is, it has the default Automatic Release), use the Hold button to put the active call on hold; then select a new loop to place a new call.

Dual Console Operation

A System 25 can be equipped with up to two SLACs, which operate simultaneously when both are in service. Both consoles can receive the same types of calls, or each can be administered to receive only certain types. When one console is out of service (see Position Busy below), most calls are directed to the other. Either or both consoles can function as a message center.

Inside users can reach either attendant by dialing 0, or a particular one by dialing the attendant's PDC. DID callers can use the Attendant's private DID number or the common queue.

Position Busy

Operation of the Position Busy button by the attendant makes the console unavailable to most incoming calls from the common queue and directs the calls to another answering station. The only types of calls that are not diverted by Position Busy are Attendant DID, DGC, and PDC calls. The placing of outgoing calls is not affected. When the Position Busy condition is active, the green status lamp of the button lights steadily. Position Busy is similar to the Send All Calls feature, which is not administrable on the SLAC.

The Position Busy feature is automatically assigned to a button position (see Figure 2-4) when the console is administered. In a one-console configuration, however, the feature is enabled only if a multiline voice terminal in the system is administered as a receiver of calls from the common queue while the console is unavailable. If this is not done, the button should be reassigned to another of the permissible features.

In a two-console configuration, an attendant in Position Busy mode will be covered by the other attendant. Only one console can be in Position Busy mode at a time.

Call Types

The following types of attendant-seeking calls are sent to the common queue and then directed to an idle LOOP button at a console:

- Incoming trunk calls that are administered to ring in the queue.
- Dial Operator calls (placed from inside stations by dialing 0)
- Following calls signed in at the console
- Calls to Floating PDCs (FPDCs) that are not signed in at a specific station
- Direct Inward Dialing (DID) calls to numbers that are not assigned to specific stations
- Attendant DID calls
- Calls to the attendant's PDC
- Coverage calls for which the common queue is a covering receiver
- Returning calls.

An incoming trunk connected directly to a DGC group can also be assigned to the Attendant Queue. Calls ringing simultaneously at the DGC group and at the console will be connected to the facility that answers first.

The order in which calls are serviced is established by system administration. Each type of call is assigned a priority that determines its position in the common queue with respect to other types. System administration also establishes where the calls go. Obviously, in a one-attendant system, all calls automatically go to that attendant. If a system has two attendants, however, administration can direct calls of each type (with the exceptions noted below) to either position or to both positions. Returning calls can be directed to either console or to the specific console that originated them.

Following and Attendant PDC calls can be assigned priorities but cannot be directed to a specific attendant in a two-position system. Any trunk except types 901-902 and 1003-1008 (DID and Dial-in Tie Trunks) can be assigned a priority and be directed to a specific attendant position or to both.

Calls accessed by dialing a code [Pickup at other extensions, Trunk-Answer-from-Any-Station (TAAS) Night Service calls, and calls parked by other stations] are originated at a LOOP button and brought to the console on that same button. These calls do not enter the common queue.

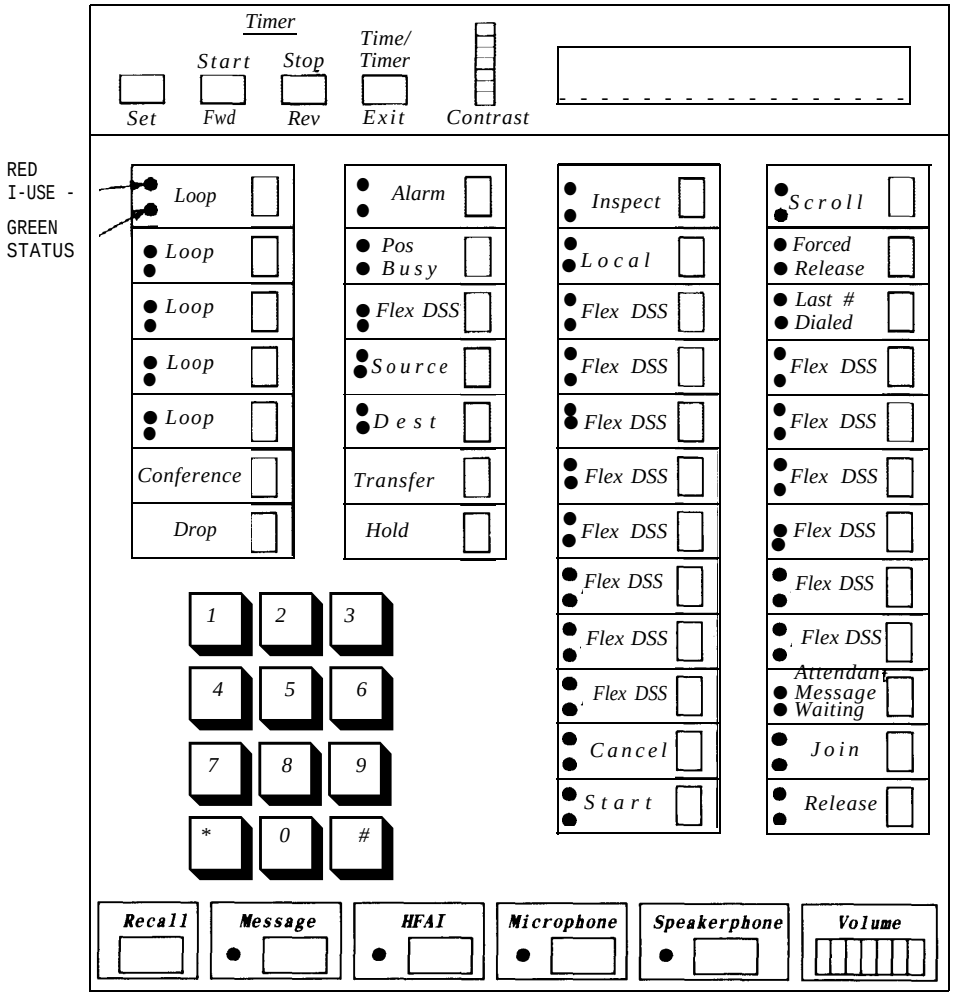


Figure 2-4. Buttons and Display of BIS-34D

Ringing

The SLAC receives normal ringing on incoming calls. Abbreviated alerting (one short burst of ringing), accompanied by a change in the LOOP button wink rate, indicates to the attendant that a held call has exceeded the preset hold time interval. Calls on hold can be administered to continue on hold after the second timeout or to return to console queue. Abbreviated alerting can also be administered as a reminder for new calls entering the queue.

Considerations

One System 25 configuration can support either one or two SLACs or one or two DTACs, but not a combination of a SLAC and a DTAC.

The optional Direct Extension Selector Console can be connected to a SLAC to provide busy/idle status of inside stations and quick calling of their extension numbers. In a system with two consoles, either or both can have a Selector Console as an adjunct. The Selector Console is covered in the "Attendant Direct Extension Selector Console" feature description.

For information on related Attendant Features (Table 2-E), refer to the individual feature descriptions.

Interactions

The following features interact with Switched Loop Attendant Console.

Attendant Console, Direct Trunk: The SLAC cannot operate in the same system with a Direct Trunk Attendant Console.

Bridging of System Access Buttons: The SLAC cannot serve as a principal or a bridging station.

Callback Queuing: The attendant can queue calls that are extended using the normal START-RELEASE button operation. However, calls originated using only the START button (no other call put on hold) cannot be queued.

Call Originations: Placing a new call from an active console causes interactions with the currently active call. At a console that has the default Automatic Release feature, the active call is dropped when the attendant presses a new LOOP button. The new loop becomes the active one; dial tone is provided, and the attendant can dial a number. If the optional Automatic Hold feature is administered, the interrupted call is put on hold instead of being lost.

If a new call is originated with the START button or at the Selector Console, the active call is split. The current loop becomes the active loop for the new call. The display shows the split call information. This is the normal operating procedure for extending calls.

When a REP DIAL button is pressed while the console is active on a call, the active party is not dropped or split, and the display does not change. If the active call is with an inside station, the digits generated by the REP DIAL button are ignored. But if the active call is on an outside trunk, then pressing REP DIAL will cause the digits stored on the button to be sent out over the trunk ("thru-dialing" or "end-to-end signaling").

When an active call is put on hold manually by operation of the HOLD button, the system does not automatically select a new loop for placing a call. In this case, the attendant can select a new loop by pressing an idle LOOP button, then dial a number. Pressing a DSS, Flex DSS, Auto Intercom, Last Number Dialed, REP DIAL, or Selector Console button will select a new loop and dial a number in a single operation.

Call Waiting: Calls cannot wait at the SLAC.

Callback Queuing: Calls that are originated without use of the START button can be queued for busy facilities. They are treated like calls from standard multiline stations. A queued call remains on the LOOP button where it was originated and does not return via the common queue.

Headset Adapter: Connection of a headset adapter to the SLAC allows the optional use of a headset instead of the handset in handling calls.

Administration Requirements

Attendant Console (Voice Terminal) Port:

- Assign voice terminal type.
- Assign Prime Line Preference to one of the LOOP buttons; default = top LOOP button.
- Assign flexible buttons.

Trunk Port:

- Assign priorities to calls directed to the console queue.
- Assign the attendant(s) to handle calls from this trunk.
- Assign unique trunk identifiers.

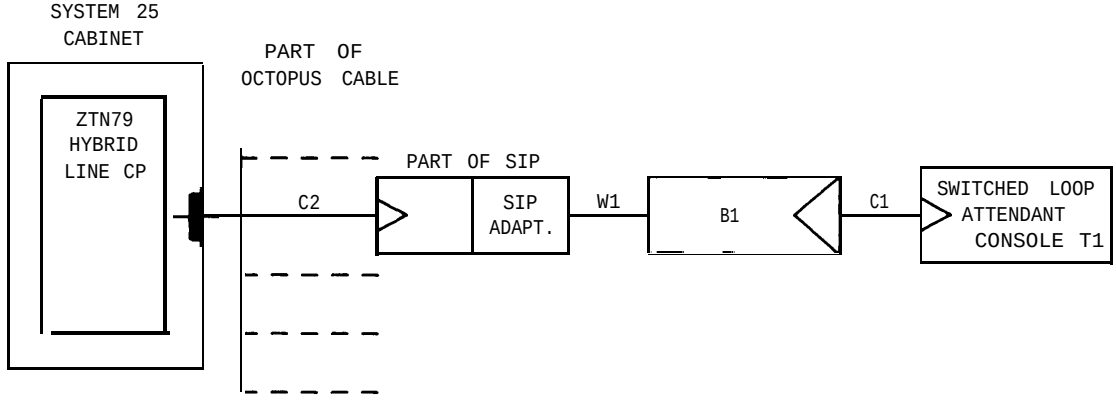
System:

- Assign DID number for attendant “0” treatment.
- Assign Coverage Group number for which the console queue is to serve as a receiver.
- Assign Automatic Hold or Automatic Release.
- Enable ring reminder when calls enter queue.
- Assign Hold timer interval.
- Assign destination of held calls that time out.
- Assign call types and attendant specification for Message Center-Like operation in a two-console configuration, if applicable.
- Assign call type priorities and attendant specification.
- Assign Position Busy “backup” station, if applicable.

Hardware Requirements

Each console requires a port on a ZTN79 ATL Station circuit pack.

Figure 2-5 provides a connection diagram for the SLAC.



LEGEND:

B1 – TYPICAL-103A CONNECTING BLOCK*
C1 – MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
T1 - 7305H04C BUILT-IN SPEAKERPHONE (BIS) WITH DISPLAY VOICE TERMINAL - PEC 3162-DIS
OR 7317H01A DELUXE BIS WITH DISPLAY VOICE TERMINAL – PEC 3167-DSB
W1 – 4-PAIR INSIDE WIRING CABLE*

* –FURNISHED BY INSTALLER

RANGE: WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED >1000 FEET)

Figure 2-5. Switched Loop Attendant Console Connections

Attendant Direct Extension Selection

Description

This feature permits the attendant to extend calls to stations by pressing one or two buttons instead of pressing START and dialing the extension number. Each attendant console in a system can have an associated Direct Extension Selector Console. A Selector Console can be used with either a Direct Trunk Attendant Console (DTAC) or with a Switched Loop Attendant Console (SLAC).

The Selector Console is also used by the attendant for simply calling inside stations, in addition to "extending" calls.

The Selector Console (Figure 2-6) has an array of 100 Direct Extension Selection (DXS) buttons plus seven Group Select buttons and a Test button. The DXS buttons are labeled 00 through 99. Default assignments for the Group Select buttons are 200-299, 300-399, etc., up to 800-899, but they can be assigned any hundreds group in the dialing plan. To select an actual extension number, the user presses a Group Select button for the hundreds group and a DXS button for the last two digits.

Pressing a DXS button when off-hook on an incoming call is equivalent to pressing START and dialing a station. Such action will busy out the Start facility until the call is released. The Selector Console can be used to monitor the on-hook/off-hook status of stations in the system. If the attendant, while on-hook, presses a Group Select button, the Group Select LED and the LEDs of any busy stations in that group will light steadily.

The DXS button LED and the Group Select button LED associated with a particular station will flash when one of the following events occurs:

- The station calls the attendant
- A call extended by the attendant to the station returns on a RTN-BUSY or RTN-DA button (DTAC only) or on a LOOP button (SLAC only)
- The station is covered and a call to it is redirected to a COVER button (DTAC only) or to a LOOP button (SLAC only).

The LEDs stop flashing when the call is answered. When the attendant answers a returning call, the LEDs will return to the state that reflects the current on-hook/off-hook status of the station. In all of the above cases, the Group Select lamp associated with the current "hundreds page" remains lighted steadily.

An outside call can be parked via the Selector Console by pressing one of the eight DXS buttons that can be designated as Park extension numbers. On the DTAC, the status LED of the parked call winks (to indicate that the call is on hold) and the status LED on the Selector Console lights steadily. On the SLAC, the call is removed from the attendant console, with the Selector Console LED lit steadily.

A call parked via the Selector Console and not picked up within an administered period (0-240 seconds; default = 120) will return to the console. The status LED of the parked-on button will flash while the call is ringing the attendant.

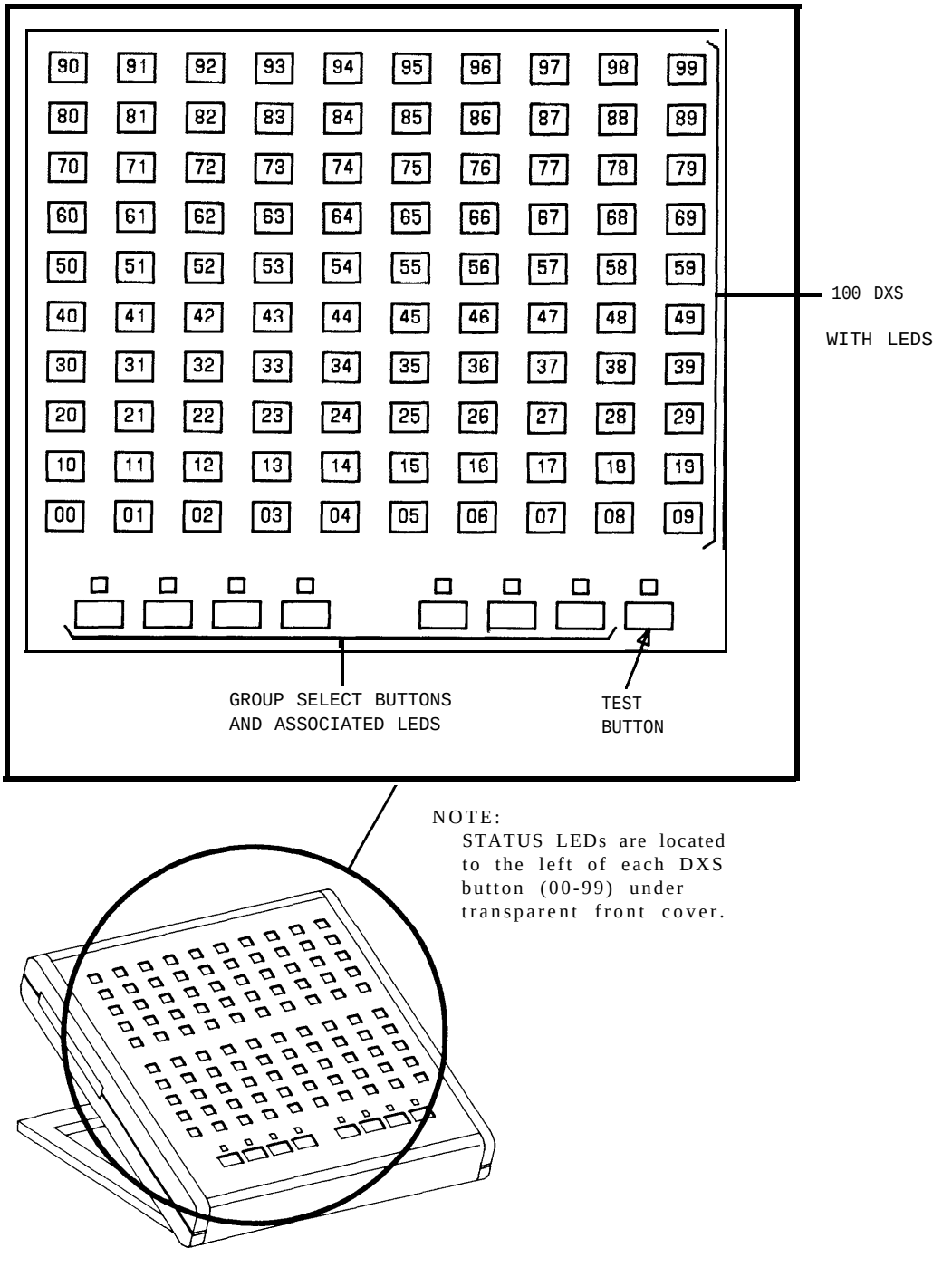


Figure 2-6. Model 23A1 Attendant Direct Extension Selector Console

A call parked via the Selector Console can be picked up at any voice terminal by dialing the Park retrieval code (*8) and the number on which the call is parked.

The rightmost button on the bottom row is a Test button. When it is pressed, all DXS LEDs will light sequentially; a second press allows individual LEDs to be tested and a third press ends the test.

Considerations

When there are two Attendant Selector Consoles in the system the Group Select button assignments are identical. Whenever an administrative change is made to one console, the other console is automatically changed.

Buttons on the Selector Console point to either station PDCs, FPDCs (FPDCs), Park codes, DGC access codes, or pooled facilities. Calls extended by the Selector Console are directed as described in the “Personal Dial Codes” feature description.

When a station calls the attendant, the associated LED on the Selector Console will flash while the call is ringing and will light steadily when the attendant answers the call. The LED will light steadily whenever the terminal is off-hook. Station busy indication is not provided for buttons pointing to FPDCs.

If a call to a PDC is directed to a COVER or LOOP button on the Attendant Console, the covered status LED of the voice terminal on the Selector Console will flash and then go dark when the call is answered by the attendant. If the covered call was intended for a FPDC that was signed in at a terminal with attendant coverage, the Selector Console status LED associated with the FPDC (if assigned) will flash. In this case, the Cover button status LED will also flash (DTAC only).

A call can arrive at an Attendant Console SYSTEM ACCESS or LOOP button because the PDC or FPDC is signed in at the Console or because the FPDC is not signed in anywhere. For these calls, the status LED on the Selector Console will not light.

If the attendant extends a call to a station or DGC group and that call returns to the attendant, the status LED of the called station or group on the Selector Console will flash and then go dark when the call is answered by the attendant. This is true regardless of the sign-in status of the PDC.

Interactions

The following features interact with Attendant Direct Extension Selection.

Attendant Position Busy: The Selector Console functions normally when the associated Attendant Console is in the inactive mode.

Attendant Return-Coverage-On-Busy/On-Don’t-Answer: If a call to a FPDC is returned to the attendant on a RTN-BUSY or RTN-DA button or on a LOOP button, the status LED of the FPDC on the Selector Console will flash during ringing and go dark when answered.

Callback Queuing: Callbacks to the attendant do not flash at the associated Selector Console.

Coverage: If the attendant receives a coverage call for a FPDC, the associated status LED on the Selector Console will flash and then go dark when the call is answered by the attendant.

Direct Group Calling: When all stations in a DGC group are busy, the DXS status LED on the Selector Console lights.

Pooled Facilities: If a 1- or 2-digit FAC is used, the associated status LED on the Selector Console will light steadily whenever all trunks in this group are busy. This does not occur with 3- or 4-digit FACs.

Administration Requirements

Special Feature Ports:

- Assign a port on a ZTN79 ATL Station Circuit Pack for each Selector Console.
- Assign Group Select button hundreds groups.

System:

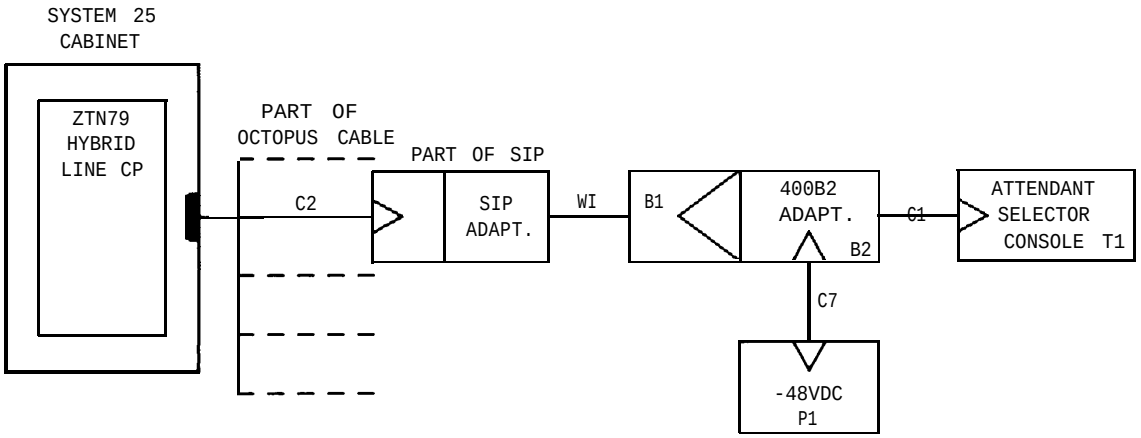
- Assign Selector Console Park codes.
- Park return time (0-240 seconds; default = 120).

Hardware Requirements

Requires an Attendant Selector Console, and a port interface on a ZTN79 ATL Station CP.

The Selector Console requires a KS-22911, List 1 Power Supply, associated 115V ac power outlet, and a 400 B-type Adapter. The 40062 Adapter provides power to the console at the wall jack. The Console connects to a port on the ZTN79 ATL Station CP.

Detailed connection information is provided in Figure 2-7. Descriptions of the Station Interconnect Panel (SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under “Connectivity” in Section 4.



LEGEND :

B1 – TYPICAL-103A CONNECTING BLOCK*
B2 – 400B2 ADAPTER - FURNISHED WITH CONSOLE
C1 – MODULAR CORD (D8W-87) - FURNISHED WITH CONSOLE
C2 – OCTOPUS CABLE (WP90780) - PEC 2720-05P
C7 – CORD D6AP-87- FURNISHED WITH CONSOLE
P1 – KS-22911 POWER SUPPLY - FURNISHED WITH CONSOLE
T1 – 23A1 SELECTOR CONSOLE - PEC 62509
W1 – 4-PAIR INSIDE WIRING CABLE*

* – FURNISHED BY INSTALLER

RANGE : WITHIN 2000 FEET OF SYSTEM CABINET
(LOCAL POWER REQUIRED >1000 FEET)

Figure 2-7. Attendant Direct Extension Selector Console Connections

Attendant Forced Release (SLAC Only)

Description

This feature drops all active parties from a call in which the attendant and one or more other parties are connected together. The attendant uses the feature by pressing the FORCED RELEASE button while connected to other callers in a conference-type call. The other parties will be disconnected from the console and from each other. After Forced Release has taken place, the attendant can receive a new call from the console queue or place a call.

Considerations

Forced Release differs from Release in an important respect. Simple Release separates the attendant from an extended call or a conference call, but leaves the other parties connected together; Forced Release completely disconnects all parties.

When the attendant is connected to only one other party, Forced Release has the same result as Release (that is, the call is ended). If the attendant has already Released, Forced Release has no additional effect.

For information on related Attendant Features (Table 2-E), refer to the individual feature descriptions.

Attendant Join (SLAC Only)

Description

This feature allows the attendant, while extending an incoming call, to connect together the calling party, the called party, and the console in a 3-way call. All parties can talk to each other. The attendant activates the feature by pressing the JOIN button while still connected to one of the other parties.

Considerations

The Join feature cancels a split condition.

The attendant can use the Join feature to stay on an extended call and give assistance to the other parties. A joined call can be expanded into a conference call by adding more parties.

Once the JOIN button has been pressed, there is no way to “unjoin” the calling and called parties (that is, separate them back into SOURCE and DEST).

For information on related Attendant Features (Table 2-E), refer to the individual feature descriptions.

Attendant Message Waiting

Description

This feature allows the attendant to remotely control the status of Message LEDs on user stations. The attendant can activate the Message LED of the station while either (1) ringing, (2) receiving Busy Tone, or (3) talking to a station. The status of the Message LED of the called party is reflected by the green status LED of the Attendant Message Waiting button in any of these cases.

To activate (light) a user's Message LED in any of these cases, the attendant presses the Attendant Message Waiting button. (If the signaled voice terminal is not equipped with a Message LED, the attendant's LED will remain dark.)

Note: The Attendant Message Waiting button on the Direct Trunk Attendant Console (DTAC) is labeled ATT MSG. On the Switched Loop Attendant Console (SLAC), the name is completely spelled out.

If the attendant presses the button a second (or third) time before hanging up, the user's Message LED will turn Off (and back On), etc.

The red I-Use LED associated with the Attendant Message Waiting button on the DTAC is inoperative.

The attendant can turn a user's Message LED on or off without disturbing the user by going off-hook on a System Access or Loop button, pressing the Attendant Message Waiting button, and then dialing the station. Confirmation tone is returned, and the user's Message LED will turn on. To turn it off, press the button again.

Considerations

This feature allows the attendant to notify stations that a message is available for them.

This feature is not the same as the Station-To-Station Message Waiting or the Coverage Message Waiting features. Refer to the "Messaging Services" feature description for a summary of all system Messaging Services.

Interactions

The following features interact with Attendant Message Waiting.

Callback Queuing: An attendant active on a queued inside call can toggle the Message LED of the queued-for station by pressing the Attendant Message Waiting button.

Conference: Pressing ATT MSG while on a conference call will be ignored.

Coverage Calls: The attendant can light the Message LED of the covered station when receiving a coverage call for the station.

Hands-Free Answer On Intercom: If the attendant lights the Message LED on a voice terminal with AUTO ANS button active, the auto-answer function will turn off, allowing subsequent calls to receive coverage as assigned.

Administration Requirements

Attendant Console (Voice Terminal) Port:

- The ATT MSG button is defaulted and fixed on SLACs; it is defaulted on DTACs but can be assigned to any programmable button.

Hardware Requirements

Stations must have a Message indicator (not assignable).

Attendant Position Busy

Description

This feature allows an Attendant Console to be placed in an inactive mode.

Systems with Direct Trunk Attendant Console(s) (DTAC)

There must be two Attendant Consoles in the system before this feature can be activated. A Position Busy (POS BUSY) button can be assigned on each of the consoles. Pressing POS BUSY at one of two active consoles causes the POS BUSY status LED to light and the console to be placed in the inactive mode. Pressing POS BUSY a second time causes the LED to go dark and the console to be reactivated. Pressing POS BUSY when only one Attendant Console is active is ignored (i.e., only one console is allowed to be inactive at a time.)

When a console is in the inactive mode, ringing is disabled on facility appearances on the two rightmost button columns *only*. The (green) status LEDs will continue to operate normally. Calls to floating PDCs not signed in, DID calls, and dial "0" calls will be transferred to the active console. Internal calls to the PDC of the inactive console will still be directed to that console.

Incoming calls on lines that normally ring at only the inactive console will now ring at the active console if they have an appearance there.

All buttons on the inactive console will continue to function normally, including the Selector Console buttons. Calls can be originated by the inactive console. Call appearances in the leftmost two columns of buttons on the inactive console are not affected by the Position Busy feature.

The attendant can press a Direct Station Selection (DSS), Automatic Intercom (AUTO ICOM), or a Direct Facility Access (FACILITY) button and then receive busy-to-idle reminder when the facility becomes idle.

Note that if a personal trunk appears on only one DTAC, incoming calls on those trunks will not receive service when the console is inactive. For this reason, it is strongly recommended that each DTAC attendant be assigned a Coverage-Individual (COVER-IND) button for the other console so that these calls can be covered. Also, be sure to make the Attendant Console the principal station (owner) on all trunks that are to receive coverage by the other attendant.

Systems with Switched Loop Attendant Console(s) (SLAC)

In order for the Position Busy feature to be operational, the system must have either two attendant positions or one position plus a multiline voice terminal administered as a "backup." If the system has two consoles, pressing the Position Busy button on one will make it inactive and cause most calls in the common console queue to be directed to the active console. Each attendant covers for the other. Only one console can be in Position Busy condition at a time. If the system has one console with an administered *backup* voice terminal, pressing the POS BUSY button will make the SLAC inactive, and most calls from the common queue will be directed to the *backup* terminal.

A console in Position Busy mode can receive attendant PDC, DID, and DGC calls, and outgoing calls can still be placed. Local functions can be activated.

Considerations

Position Busy allows one of two attendant positions to be made inactive when not required. This is useful in situations where calling traffic requires only one console operator.

Interactions

The following features interact with Attendant Position Busy.

Attendant Call Extending: Unanswered calls extended by an inactive console will return to the active console on the Return-On-Don't-Answer (RTN-DA) button (DTAC only) or on a Loop button (SLAC only).

Attendant Camp-On: Calls Camped-On by an inactive console will return to the active console when Camp-On timeout occurs.

Attendant Message Waiting: An inactive attendant is permitted to control voice terminal Message LEDs.

Automatic Intercom: The inactive attendant is permitted to place Automatic Intercom calls. DTAC only: Automatic Intercom calls to the inactive attendant will not ring at the console or be transferred to the active attendant when the AUTO ICOM button is located in one of the two rightmost button columns

Coverage:

- DTAC only—If the active attendant is a coverage receiver for the inactive attendant, coverage is invoked and calls will appear at the active attendant's Cover button. If the inactive attendant is a coverage receiver for the active attendant, coverage, when activated, is invoked at all coverage stations including the inactive attendant. However, if the Cover button is located in one of the two rightmost button columns, coverage calls will not ring at these buttons.
- SLAC only—All calls covered by the common queue will be directed to the active console.

Direct Group Calling: If the inactive attendant is a member of a DGC Group, calls directed to the group will be routed to the inactive attendant. The attendant must dial *4 (activate DGC Group "Make Busy") to busy out from the group. Dialing *6 deactivates the "Make Busy" function.

Direct Inward Dialing: All DID calls to unassigned DID numbers will be transferred to the active attendant.

Forwarding: If a PDC/FPDC is signed in at an inactive attendant console, then calls to this PDC/FPDC will go to the active attendant (SLAC) or to the inactive attendant (DTAC only). All calls to FPDCs not signed in will be transferred to the active attendant.

Night Service: An inactive attendant that is a Directed Night Service receiver will receive Night Service calls.

Park: A call parked on the inactive attendant console will return to the inactive console if the call times out; calls parked via the Selector Console will return to the active console.

Personal Lines: All calls to trunks having an appearance in either of the two leftmost button columns of a DTAC will ring normally at the inactive console. All calls to trunks having appearances in either of the two rightmost button columns will not ring. If these trunks also have an appearance at the active console, they will ring there even if they do not normally.

Program: The Program feature remains active at the inactive console.

Programmable Buttons: All DSS, Flex DSS, and REP DIAL buttons remain active on the inactive console.

Administration Requirements

Attendant Console (Voice Terminal) Port:

- Assign Position Busy button on DTAC. A Position Busy button is defaulted to the SLAC; it can be assigned to another feature if desired.
- Assign COVER-IND buttons between consoles (DTAC only).

System

- Designate “backup” multiline voice terminal (single-SLAC systems only).

Attendant Release

Description

This feature releases the attendant from an extended call. There are two forms of Attendant Release: Manual and Automatic. This feature applies to the Direct Trunk Attendant Console (DTAC) and the Switched Loop Attendant Console (SLAC).

Manual Release:

Pressing RELEASE releases the attendant from an extended call and completes the associated call transfer. The status LED of the original calling facility will change from hold to busy for direct trunk terminations and from hold to idle for other call facilities {e.g., Loop, Return-On-Busy, Return-On-Don't-Answer, Cover, Automatic Intercom, DSS, and System Access).

Calls cannot be released to Reorder or Dial Tone.

Pressing CANCEL terminates the destination call and reconnects the attendant to the calling party. If the attendant goes on-hook without first releasing a call, the call extending operation will be terminated (the calling party will be disconnected).

Automatic Release:

This feature simplifies the attendant procedures by eliminating the need for the attendant to press RELEASE when releasing from one call to handle another. Selection of any new line facility while active on the Start button will automatically release the first call. At release, the status LED of the first calling facility will change from hold to busy for direct trunk terminations and from hold to idle for other call facilities (e.g., Loop, Return-On-Busy, Return-On-Don't-Answer, Cover, Automatic Intercom, DSS, and System Access).

The Automatic Hold feature can be administered for the SLAC as an alternative to Automatic Release. if the attendant, active on a loop call, presses another loop button to place a call or pick up a held call, the active call is put on hold—not released.

Considerations

Attendant Manual Release improves attendant efficiency in handling calls by allowing the attendant to release an extended call without having to wait for the called station to answer.

Attendant Automatic Release enhances the attendant's ability to handle many calls by eliminating the Release operation when answering a second call.

The Release function is inhibited whenever the Start facility is connected to Reorder or Dial Tone. Pressing CANCEL will terminate the destination call and reconnect the attendant to the calling party.

Administering the Automatic Hold option instead of Automatic Release reduces the occurrence of accidentally dropped calls.

Interactions

The following feature interacts with Attendant Release.

Attendant Camp-On: External calls that are released when Busy Tone is heard will be camped on.

Administration Requirements

System with SLAC: Enable Automatic Hold feature? (yes for Automatic Hold or no for Automatic Release; default = no).

Attendant Return Coverage On Busy

Description

This feature allows a camped-on call at a busy station or DGC Group to be returned to the attendant for service after a specified time period.

A camped-on call not answered within 1 to 120 seconds (administrable) after the attendant releases the call will return to the console in one of the following ways:

- On the Return-On-Busy (RTN-BUSY) button at a Direct Trunk Attendant Console (DTAC).
- On a LOOP button at a Switched Loop Attendant Console (SLAC).

To answer a returned call at a DTAC, the attendant presses RTN-BUSY (if not selected by Ringing Line Preference.) A returned call can be reextended via the START button or a Selector Console button. In either case, the Return-On-Busy button is idled as soon as the attendant releases.

To answer a returned call at a SLAC, the attendant merely lifts the handset to be connected to the ringing loop.

When the RTN-BUSY button is busy at a DTAC, the calling party will remain on-hold; if a loop is not available at a SLAC, the returning call remains in the console queue. The system will continue to attempt to ring the called station until the RTN-BUSY button is idle or a loop is open. When Attendant Camp-On is not provided (Camp-On return time set to zero seconds), calls released by the attendant to busy tone are returned to the console immediately.

Considerations

Attendant Return-Coverage-On-Busy allows the attendant to service calls not answered within specified time intervals. This provides the calling party better service, and results in fewer lost calls.

Interactions

The following features interact with Attendant Return Coverage On Busy.

Attendant Camp-On: External calls that are released when Busy Tone is heard will be camped on.

Attendant Console, Direct Trunk: As long as an Attendant Console remains active, the call will return to the attendant who extended it.

Attendant Console, Switched Loop: **A** returning call is directed from the common queue to a LOOP button. In a two-console system, returning calls can be administered to go to the first attendant, the second attendant, either attendant, or to the specific attendant who originated the call.

Attendant Direct Extension Selection: If a call to a Floating PDC (FPDC) is returned to the attendant, the FPDC's status LED on the Selector Console will flash during ringing and go dark when the call is answered.

Direct Group Calling: External calls that are camped onto a DGC group that does not have a delay announcement will return to the attendant console after the specified number of rings.

Send All Calls DTAC only): If Send All Calls is activated, returning calls will ring at the DTAC.

Administration Requirements

System:

- Assign number of seconds before unanswered camped-on calls return to the Attendant Position (1-120 or 0 for no Camp-On; default = 30),

Attendant Return Coverage On Don't Answer

Description

This feature allows unanswered calls extended by the attendant to be returned to the attendant for additional service.

Calls that are not answered after a administered number of rings will transfer ringing to the Return-On-Don't-Answer (RTN-DA) button on a Direct Trunk Attendant Console (DTAC) or to a LOOP button on a Switched Loop Attendant Console (SLAC). If the called voice terminal has Coverage, the counting of rings for return begins only after the coverage station begins ringing. If the terminal does not have Coverage but does have delayed ringing on System Access or Bridged Access buttons, the delay interval (administered as an equivalent number of rings) must expire before counting begins.

When the RTN-DA button is busy, calls will continue to ring at the called station until the button is idle. If a SLAC is not available to receive the returning call, it stays in the common queue until it can be serviced.

To answer a returned call at a DTAC, the attendant presses RTN-DA (if not selected by Ringing Line Preference.) The call can be reextended via the START button or Selector Console. In either case, the RTN-DA button is idled as soon as the attendant releases.

To answer a returned call at a SLAC, the attendant merely lifts the handset to be connected to the ringing loop.

Considerations

Attendant Return-Coverage-On-Don't-Answer allows the attendant to service calls not answered within a specified number of rings. This provides the calling party better service and results in fewer lost calls.

Interactions

The following features interact with Attendant Return Coverage on Don't Answer.

Attendant Console, Direct Trunk: As long as an Attendant Console remains active, the call will return to the attendant who extended it.

Attendant Console, Switched Loop: **A** returning call is directed from the common queue to a LOOP button. In a two-console system, returning calls can be administered to go to the first attendant, the second attendant, either attendant, or to the specific attendant who originated the call.

Attendant Direct Extension Selection: If a call to a Floating PDC (FPDC) is returned to the attendant, the FPDC's status LED on the Selector Console will flash during ringing and go dark when the call is answered.

Coverage:

- DTAC only—Whenever a DTAC attendant is a coverage receiver for a particular coverage group and a call is placed from the attendant position via the Start button or the Selector Console to a voice terminal in that group, the Coverage-Group (COVER-GRP) button on the Attendant Console will not track the call (COVER-GRP button status LED will not flash). If the call remains unanswered, it will return to the Attendant Console on the RTN-DA button rather than the COVER-GRP button.

- SLAC only—Whenever the common queue is a receiver for a coverage group and a call is placed from the attendant position via the START button or the Selector Console to a voice terminal in that group, an unanswered call will return to the attendant queue as a Return-On-Don't-Answer call (instead of as a coverage call)

Send All Calls (DTAC only): If Send All Calls is activated, returning calls *will* ring at the DTAC.

Administration Requirements

System:

- Assign number of rings before call return to the Attendant Position (1-31; default = 5).

Attendant Source and Destination (SLAC Only)

Description

This feature allows the attendant, while extending a call, to switch back and forth between the calling party (the source) and the called party (the destination) before connecting them together.

Pressing the SOURCE button on the SLAC after the called party has been reached has these results:

- The called party (the destination) is put on hold.
- The attendant is reconnected to the calling party (the source).
- The green status lamp of the DEST button starts winking to indicate that the destination is on hold.
- The green status lamp of the SOURCE button goes from winking to dark.

Pressing the DEST button after the source has been reconnected has these results:

- The source is put on hold (again).
- The attendant is reconnected to the destination.
- The green status lamp of the SOURCE button starts winking to indicate that the source is on hold.
- The green status lamp of the DEST button goes from winking to dark.

Considerations

The Source/Destination feature is useful when the attendant needs to talk to each party privately before connecting them.

Interactions

The Source/Destination feature can only be activated *before* the two parties are connected together.

When the attendant presses JOIN, the other parties and the attendant are joined in a 3-way connection. When the attendant presses RELEASE, the other parties are connected, the call is separated from the console, and the attendant is free to handle other calls. After the source and destination parties are connected together, the SOURCE and DEST status lamps go dark.

Attendant Splitting One-Way Automatic

Description

This feature allows the attendant to converse privately with a called party while the calling party is split away on hold.

When the attendant presses START (or a DXS button) to extend an incoming call to a called party, the calling party is automatically split away from the connection and placed on hold. This allows the attendant to talk privately with the called party before extending the call. The attendant can then press RELEASE to complete the transfer or CANCEL to drop the called station and return to the incoming call.

If the console is a SLAC, the attendant can also use the Join and Source/Destination features while in the Start mode. Refer to the descriptions of these features for details.

Considerations

Attendant Splitting One-Way Automatic allows the attendant to (1) announce a call, (2) determine privately whether the called party is available to receive the call, and (3) obtain information if necessary to redirect the call or take a message.

Interactions

The following features interact with Attendant Splitting One-Way Automatic.

Attendant Source/Destination (SLAC only): This feature can be used after reaching the called party. It allows the attendant to speak privately to both the calling party and the called party before connecting them together.

Music-on-Hold: Music-on-hold *is not* provided to the calling party while the call is split from the console.

Attendant System Alarm Indication

Description

This feature provides a visible alarm on the Attendant Console to alert the attendant to problems detected by the system software. The ALARM LED on the Attendant Console will flash whenever a detected fault persists longer than four minutes, or if more than five transient faults per hour are detected. *The alarm indication should be reported immediately to your AT&T Systems Technician.*

The alarm type that causes an alarm indication is referred to as a Permanent System Alarm. These alarms are faults that can cause degradation of service and require immediate attention.

If a flashing ALARM button is pressed, the LED will change from flashing to steadily lit. A new trouble situation will cause a steady ALARM LED to start flashing again. Only when the trouble has been corrected will the LED turn off.

Considerations

The ALARM LED on the Attendant Console provides a warning as soon as the fault is detected. This permits a quick response to system detected faults.

In a two-attendant system, both consoles track problems.

Automatic Intercom

Description

This feature allows a multiline voice terminal user (or attendant) to place and answer calls to and from another station by use of a dedicated button appearance.

Automatic Intercom provides a private path between two designated multiline voice terminals. To place an Automatic Intercom call, the calling party presses the Automatic Intercom (AUTO ICOM) button and goes off-hook. The calling party hears ringback tone and the called party receives standard ringing. The status LED associated with the button is steadily lit at the calling voice terminal and flashing at the called voice terminal. To answer an Automatic Intercom call, the called party presses AUTO ICOM (not necessary with Ringing Line Preference) and goes off-hook.

The AUTO ICOM status LED lights steadily whenever the other party is off-hook. This provides each party with a station-busy indication for the other. To activate the busy-to-idle reminder, the user can press AUTO ICOM (remaining on-hook). A short burst of tone is provided when the other party goes on-hook. The user can then go off-hook, and the call will be placed; the user does not press the AUTO ICOM button again.

Pressing AUTO ICOM to invoke the busy-to-idle reminder overrides Prime Line Preference. Once activated, the feature can only be canceled by preelection of another button or answering an incoming call. See the “Busy-to-Idle Reminder” feature description for more information.

At a Switched Loop Attendant Console, operation of an AUTO ICOM button seizes an idle loop button for the outgoing call. An incoming Automatic Intercom call arrives on a loop button and does not flash at the AUTO ICOM button of the console.

Considerations

The intercom feature should not be confused with ordinary station-to-station calling inside the system using dialed PDCs. With Automatic Intercom, users who frequently call each other can do so by pressing one button instead of dialing a PDC. In addition, the station-busy indication and busy-to-idle reminder provide additional utility to users.

This feature is similar to Direct Station Selection (DSS), except that the buttons must always be assigned in pairs (i.e., between two sets.) Hence, an AUTO ICOM button cannot point to a single-line set. Also, Automatic Intercom calls arrive at the AUTO ICOM button, thereby providing calling party ID; DSS calls arrive on System Access buttons.

Interactions

The following features interact with Automatic Intercom.

Attendant Position Busy: The inactive attendant is permitted to place Automatic Intercom calls. Automatic Intercom calls to an inactive DTAC where the AUTO ICOM button is located in one of the two rightmost button columns will not ring at the console, nor can they be covered by the active attendant. However, Automatic Intercom calls to an inactive SLAC *will* ring there.

Bridging of System Access Buttons: Calls on Automatic Intercom buttons on the principal station are not accessible from bridged call appearances on the bridging station.

Coverage: Automatic Intercom calls are considered private and do not receive coverage.

Direct Group Calling: Automatic Intercom calls cannot be directed to DGC groups.

Exclusion: Any attempt to engage Exclusion while active on an Automatic Intercom call will drop the other party.

Last Number Dialed: Numbers called using an AUTO ICOM button are not saved by the Last Number Dialed feature.

Line Selection (Prime Line Preference): When the Automatic Intercom line is assigned Prime Line status, the AUTO ICOM button must be pressed to activate the busy-to-idle reminder even though the I-use LED is already lighted steadily.

Pickup: When an Automatic Intercom call is answered via the Pickup feature, the AUTO ICOM status LED on the called voice terminal lights steadily. The called party can press AUTO ICOM to enter the call at any time.

Administration Requirements

Voice Terminal Port:

- Assign AUTO ICOM buttons to voice terminals in pairs. Voice terminals can have several AUTO ICOM buttons assigned for direct access to multiple stations.

Automatic Route Selection (ARS)

Description

This feature provides for the routing of calls over the telecommunications network based on preferred routes (normally the least expensive route available at the time the call is placed.)

An ARS pattern can be composed of two subpatterns (time of day determines which subpattern is selected), each consisting of up to three routes, associated Facility Restriction Level (FRL) codes (described below), and CO overflow flags. A route is identified by specifying a Facility Access Code for the pooled facility (trunk group); a route may also be identified by specifying a Virtual Facility code.

A trunk group or virtual facility can be used in more than one ARS pattern and more than once within a pattern.

Call routing can be specified by as many as eight routing patterns. Each pattern contains a sequential list of routes (for example, trunk groups) the system can use to complete a call. Number translations (deletion and addition of dialed digits) necessary to route the call is determined on a trunk group basis. Overflow to the local CO when all trunks in a pattern are busy or the route FRL is too high is optional. If all trunks in a pattern are busy (including CO trunks if overflow is allowed), the call may be queued (via the Callback Queuing feature) on the first route in the pattern.

All calls placed using the ARS access code (default = 9) are routed via the feature. The dialed numbers that follow the ARS access code are generally seven or ten digit DDD numbers preceded by a " 1'' if required by the serving Central Office. Numbers preceded by a "0" are routed over the local CO pooled facility.

Typically, a dialed 7-digit number consists of a CO code and exchange number in the form NXX-YYYY where N = 2-9, X = 0-9, and Y = 0-9. A 10-digit number consists of an area code, CO code, and exchange number in the form NPA-NXX-YYYY where N = 2-9, P = 0-1, A = 1-9, X = 0-9, and Y = 0-9.

Each route in a pattern has an associated FRL (0-3), This FRL may differ each time the facility is specified as a route. A facility with a FRL of "0" is least restricted to callers; a FRL of "3" is the most restricted. Similarly, each station in the system is assigned an FRL (0-3). A terminal assigned an FRL of "0" has the least ARS privileges (i.e., routes with FRLs of 1-3 are restricted); a FRL of "3" provides the most privileges. To use a route, a station's FRL must be equal to or greater than the route's FRL.

The ARS feature, when accessed, selects a pattern as follows:

- Emergency Number Calls (routed via the local CO facility)
- Service Code (N11) Calls (routed via an associated routing pattern)
- International Calls (routed via the administered international pattern)

- Calls made to specified COs or seven digit telephone numbers within the Home Number Plan Area (HNPA). These calls are routed as specified in the HNPA Exception Lists, or else via the NPA Routing Table or (by default if not otherwise specified) the local CO facility.

- Calls made to NPAs outside the HNPA, sometimes referred to as Foreign NPAs (FNPAs). The route selected depends on the type of call, as follows:

  FNPA special number calls (includes all “800”, “900”, and Telex 510, 610, 710, and 810 numbers). Each FNPA of the form N00 and N10 may be assigned to a routing pattern.

  — FNPA calls made to numbers specified in the FNPA Exception List.

  — All other FNPA calls.

ARS Flow Chart

Figure 2-8 provides a simplified ARS flow chart. Bracketed numbers (e.g., [401], [601]) provide a link between ARS administrable action numbers and the associated item on the flow chart. Certain readers may find this reference useful when reading the following description in association with the *System 25 Administration Manual.* Administrable System, Station, Toll Allowed, and Trunk action numbers are also noted where applicable.

The ARS feature is accessed when a user dials the ARS access code. As shown on Figure 2-8, the number dialed is first checked against the Emergency Numbers List. This list consists of special service code 911 and up to three customer-defined seven digit numbers. If the number dialed matches one of the numbers on the list, the call is immediately routed via the local CO facility. All user call restrictions are disregarded.

If the number dialed is not on this list, a check is made to determine if the terminal is allowed to originate outside calls. If the terminal is outward restricted, the caller receives Reorder Tone; otherwise, the dialed number is checked against any toll restrictions that apply.

Terminals may be assigned a Toll Restriction Class (1-4), or be unrestricted (Class 0). Terminals assigned Toll Restriction Class 1 have the most privileges, those assigned Class 4 have the least privileges. There are four associated Toll Call Allowed Lists (1-4) in the system. Up to 64 3-digit CO codes and 6-digit NPA plus CO codes may be divided among the four lists.

Numbers dialed from voice terminals assigned Toll Restriction Class 1 are checked against all four Toll Call Allowed (TCA) Lists; numbers dialed from Class 2 terminals are checked against TCA Lists 2-4; numbers dialed from Class 3 terminals are checked against TCA Lists 3-4; and numbers dialed from Class 4 terminals are checked against List 4 only. If the number dialed does not appear on a list, the user receives Reorder Tone. Calls originated at unrestricted (Class 0) terminals are not screened.

Calls are checked to determine if they are international calls or operator calls. Dialed numbers “01” or “011” signify international calls, “0” plus a number other than “1” signify operator calls (00 calls signify lntra-Lata operator calls). If the call is an international call, the international routing pattern is selected and the call routed accordingly. Operator calls are

routed via the local CO facility.

Calls within the HNPA are checked to determine if a special N11 service code (N = 2-8) has been dialed. Dialed N11 codes assigned a routing pattern are routed via the routing pattern. All other call types are checked against the HNPA Exception Lists. There may be up to four of these lists, each with an associated ARS Routing Pattern. Up to 800 3-digit office codes may be divided among the four lists (eight entries may be 7-digit numbers.) If a match is found, the call is routed via the associated ARS Routing Pattern. If no match is found the dialed number is routed via the HNPA pattern (specified in the NPA Routing Table), or if none is specified, via the local CO facility.

If a number is entered more than once in the exception list, the pattern used will be the pattern associated with the more specific number.

The NPA Routing Table is simply a listing of North American Plan NPAs and Special Number NPAs, each having an associated ARS Routing Pattern (all North American NPAs are assigned routing pattern 1 by default). A dialed NPA that is listed in the table is routed using the associated Pattern. Calls to NPAs not listed are routed via the local CO facility.

The dialed numbers are checked against the FNPA Exception List. Up to 32 entries may be assigned to the list. Each entry must consist of a 3-digit NPA code, 3-digit CO code, and two additional digits (for a total of 8 digits). The last two digits may be ".", which match any digit. Each entry has an associated ARS Routing Pattern. if a match is found, the call is routed using this pattern. If no match is found, the call is then checked against the NPA Routing Table. A dialed NPA that is listed in the table is routed using the associated Pattern. Numbers that don't match are routed via the local CO facility.

ARS Routing Pattern Table

Figure 2-9 provides a block diagram of an ARS Routing Pattern. Up to eight of these patterns may be administered in the system. Each pattern consists of two subpatterns that may be chosen based on the time of day. Each subpattern (A and B) can contain up to three allowed routes. If all routes in a subpattern are busy, a CO overflow flag (when set) allows the call to be routed via the local CO facility; otherwise, the call will queue on the first route in the subpattern.

Administrable Start and Stop times (Hour and Minute) for Routing Subpattern A specify when Subpattern A should be used to route calls. Subpattern B is used to route calls at all other times.

Each route is specified by its trunk group facility access code or Virtual Facility code and an associated FRL.

An FRL is typically lower for the first route in a subpattern and increases with each additional route in the pattern. A terminal's FRL must be equal to or greater than the route FRL for the route to be selected. The system first checks the Route #1 for an available trunk on which to route the call. If the route is busy, Route #2 is checked, then Route #3, if required. if all routes in the subpattern are busy and the CO overflow flag is set, the voice terminal FRL is checked against an associated Overflow FRL before routing the call.

If all routes in a subpattern are busy and the CO Overflow flag is not set, or all CO trunks are busy, the call returns to the first route in the subpattern and may be queued (if the station FRL permits access to the first route) via the Callback Queuing feature. A route #1 must be specified in the subpattern for a call to queue. If it is not, callers receive Reorder Tone and will not be able to queue.

Once a route has been selected, the entries in a Digit Translations Table associated with the selected route's trunk group or Virtual Facility is checked. Based on an associated NPA and the NPA dialed, the system can remove up to 10 digits and then add a pattern of up to 5 digits as specified to route the call.

The following tones are associated with ARS:

- Confirmation—Indicates that a queued call is being serviced (trunk available to route call)
- Busy—Indicates that the called number is busy
- Reorder—Indicates that all trunks are busy or that ARS calling is denied.

Considerations

With ARS, users do not have to worry about accessing a particular pooled facility to make a long distance call. The user simply dials the ARS access code and the desired number. The system then routes the call via the facility best suited for that call.

The following provides a summary of the ARS call routing controls provided by the feature:

- **Emergency Numbers List:** 911, and up to three customer-defined, 7-digit numbers.
- **Service Codes (N11 Numbers):** An ARS Routing Pattern can be assigned to each N11 (N = 2-8) Service Code. If no routing pattern is assigned, the system assumes that the N11 number is a local CO code and will wait for four additional digits to be dialed before processing it as a local call.
- **Toll Call Allowed Lists:** 1-4 lists, 64 entries maximum between all lists. Entries may be 3-digit CO codes or 6-digit NPA plus CO codes. (Administrable for uses besides ARS)
- **Station Toll Restriction Class:** 1-4 Classes (Administrable for users besides ARS)
- **HNPA Exception List: 1-4 Lists:** each with an associated ARS Routing Pattern. 800 3-digit CO codes entries maximum between all lists (eight of the entries may be 7-digit numbers.)
- **NPA Routing Table:** Entries may include every North American NPA and Special Number NPAs, each with an associated ARS Routing Pattern. All NPAs are assigned Routing Pattern #1 by default.

- **FNPA Exception List:** One List with up to 32 eight-digit numbers. Each entry has an associated ARS Routing Pattern.
- **Digit Translations Tables:** One per trunk group or Virtual Facility.

A system can have up to eight ARS Routing Patterns assigned. Each pattern can contain up to six routes (three per subpattern).

Interactions

The following features interact with Automatic Route Selection.

Bridging of System Access Buttons: When a station user originates a call on a Bridged Access button and dials the ARS code, the call is completed according to the restrictions assigned to that station, not the principal station.

Callback Queuing: Implementation of Callback Queuing affects ARS in three respects:

- On-hook ARS queuing is allowed.
- Callback Queuing and ARS share a common queue, which has a capacity of 64 calls.
- When a station without automatic queuing originates an ARS call, the caller hears reorder tone if all routes are busy; the call can then be queued manually.

Calling Restrictions: Outward Restriction and Toll Restriction, when administered, prevent calls from routing via ARS. Pooled facility access restrictions do not apply. In fact, the recommended arrangement to ensure that users make maximum use of ARS is to block dial access to most trunk groups, so that users must dial the ARS access code to place calls.

Direct Facility Access Button: Multiline voice terminal users who have pressed FACILITY to activate the busy-to-idle reminder must wait until all queued ARS users have been serviced.

Tie Trunks: Immediate Dial tie trunks should not be used in ARS routing patterns.

Virtual Facilities: Virtual Facilities may be used in place of trunk groups in ARS routing patterns. When used with ARS, a digit translation scheme may be associated with each virtual facility. See the “Virtual Facilities” feature description for more information.

Administration Requirements

System:

- Specify your area code (HNPA).
- Specify whether “Dial 1” is needed for calls outside of your area code.
- Specify whether “Dial 1” is needed for toll calls within of your area code.

ARS:

- ARS Access Code (1-9999; default = 9).
- International ARS Routing Pattern Number (1-8; no default).
- Three Emergency Numbers Lists (7-digit numbers; no default).
- NPA Routing Table (NPA code and associated pattern number 1-8; default = 1).
- N11 Routing Table (N11 code and associated routing pattern number 1-8 or 0 if this code is a valid CO exchange requiring four additional digits to complete; no default).
- HNPA Exception List:
  - List Number (1-4).
  - Pattern Number (1-8).
  - Exception Numbers (NXX or NXX-YYYY). Last three digits may be “.”
- FNPA Exception Telephone List:
  - Pattern Number (1-8).
  - Telephone Number (8-digits in the form NPA-NXX-YY; last two digits may be “.”).
- Digit Translations Table:
  - Trunk Group Facility Access Codes (1-9999) or Virtual Facility Codes (#190-#199).
  - Associated NPA (NPA).
  - Number of digits to remove for calls within associated NPA (1-10 digits, none).
  - Digit pattern to add for calls within associated NPA (maximum of 5 digits; default = none).
  - Number of digits to remove for calls not in associated NPA (1-10, none; default = none).
  - Digit pattern to add for calls not in associated NPA (maximum of 5 digits; default = none).
- ARS Routing Pattern:
  - Pattern Number (1-8).

— Subpattern A Start and End Time (Hour: Minute).

— Subpattern A and B—Route 1, 2, and 3 Facility Access Codes (1-9999) or Virtual Facility Codes (#190-#199).

— Subpattern A and B—Route 1, 2, and 3 FRLs (0-3).

— Subpattern A and B Overflows to CO facility (yes or no).

— Subpattern A and B Overflow FRL (0-3).

Terminal Port:

- ARS FRL Level (0-3).

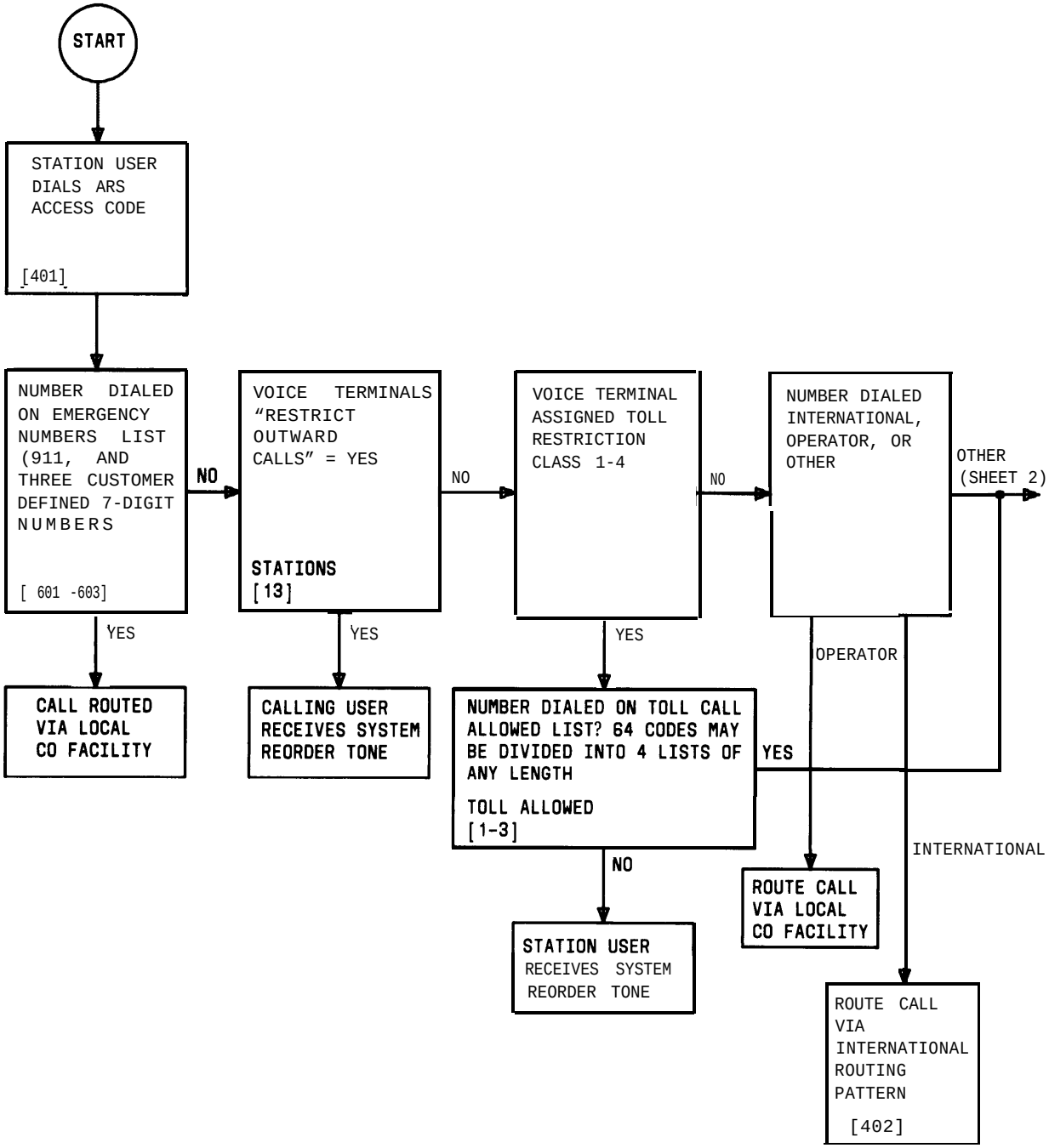


Figure 2-8. Automatic Route Selection Flow Chart (Sheet 1 of 2)

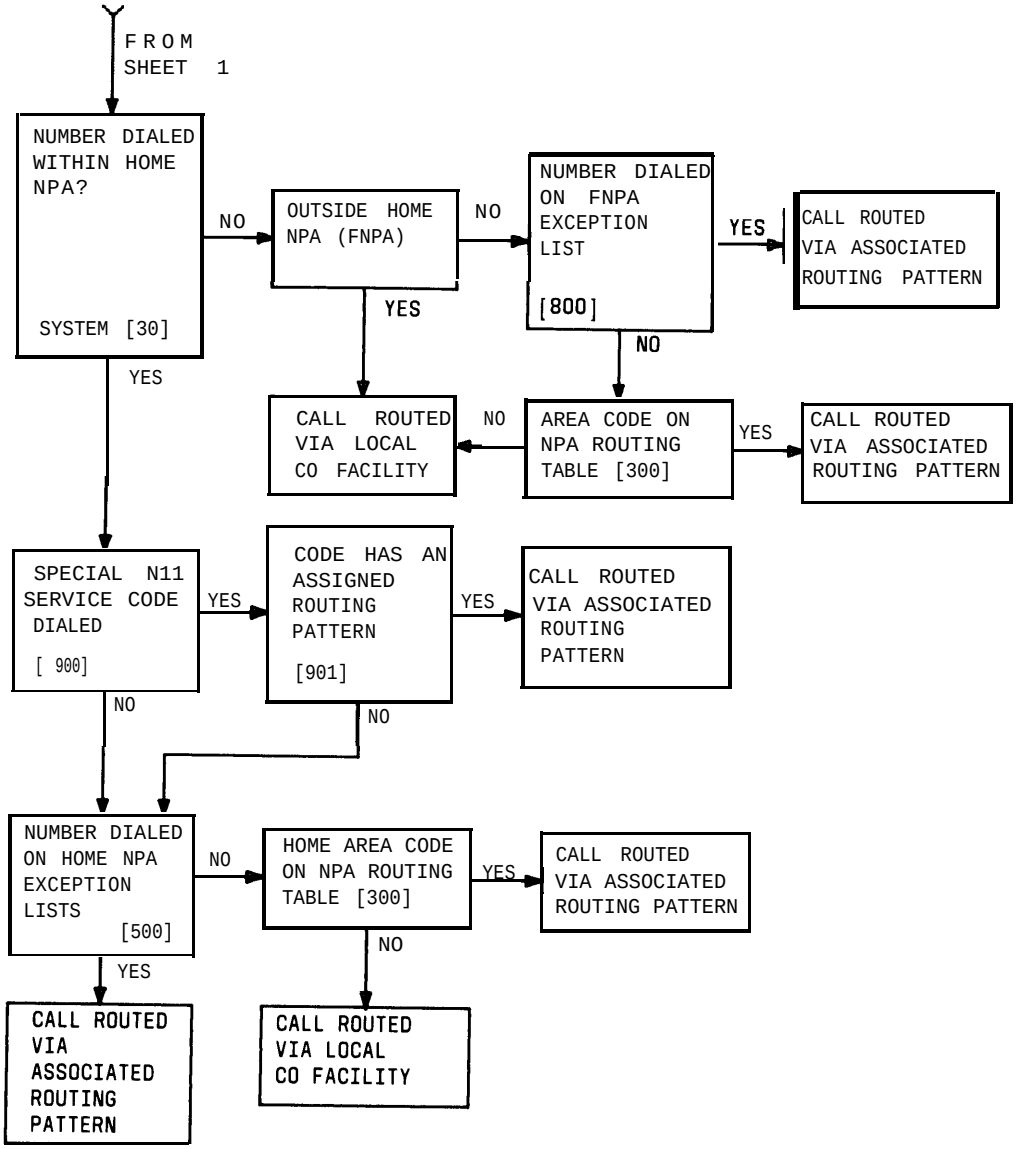


Figure 2-8. Automatic Route Selection Flow Chart (Sheet 2 of 2)

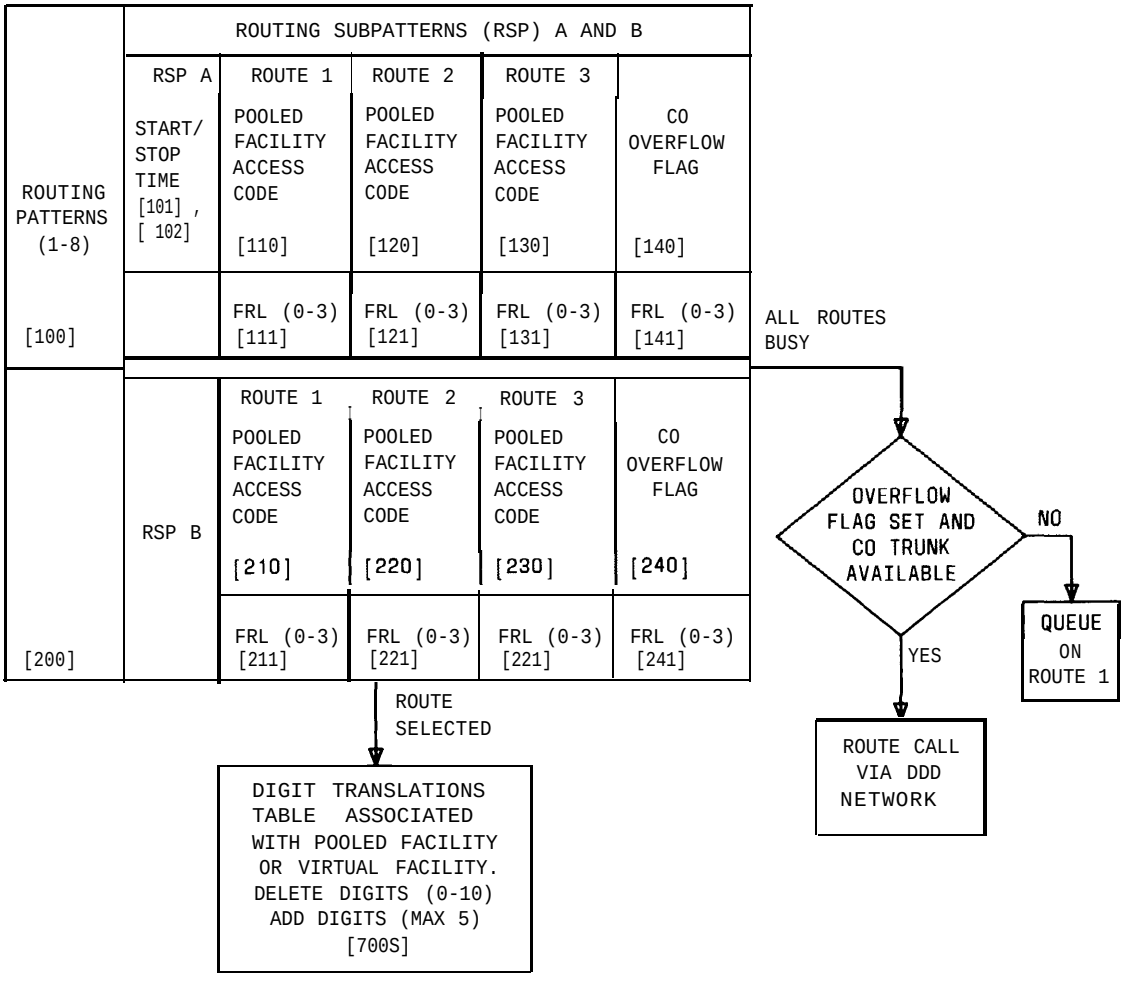


Figure 2-9. Automatic Route Selection Routing Pattern

Bridging of System Access Buttons

Description

This feature allows a multiline station user to handle calls on Bridged Access (BA) buttons associated with System Access (SA) buttons of another multiline station (Figure 2-10).

The following terms are used throughout this feature description:

- Principal station—a multiline voice terminal that has SA buttons bridged by one or more other multiline stations.
- Principal—the user of a principal station.
- BA button—a special call appearance button on a multiline station administered to correspond to a specific SA button on another user's station (the principal station); collectively referred to as bridged appearances.
- Bridging station—a multiline voice terminal with one or more BA buttons.

A BA button can be assigned to any programmable feature button on a bridging station, but it does not take the place of an SA button. The SA buttons on the bridging station can be bridged by other stations.

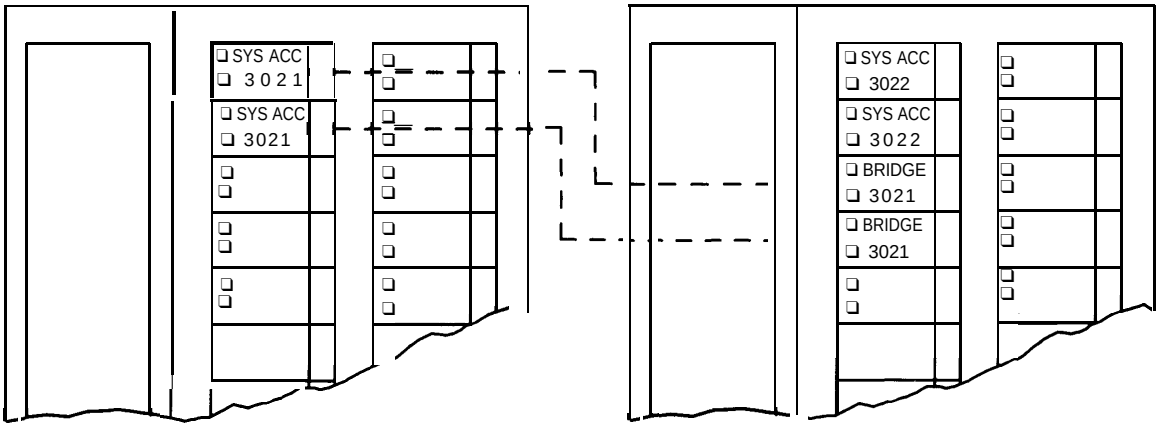


Figure 2-10. Typical Bridging Arrangement

A bridging station user is able to originate calls from BA buttons and can answer calls for the principal. The bridging user can also enter existing calls on bridged appearances by simply going off-hook on the BA button, unless Exclusion has been activated or the maximum of five active parties on a call has been reached.

The user active on a BA button can use many features with the bridged call; operation is no different from calls on other buttons. For example, Hold, Conference, and Transfer can be used from a BA button just as they would be used from an SA button. Calling restrictions applied to calls made on bridged appearances are those of the bridging station, not those of the principal station.

System 25 has the following bridging capacities and limitations:

- The number of principal stations is limited only by the number of multiline sets on the system.
- The number of SA buttons a principal station can have is limited only by the number of available buttons on the terminal.
- Each SA button on a principal station can have a bridged call appearance on up to 16 multiline voice terminals.
- The number of principals for which a station has bridged call appearances is limited only by the number of available programmable buttons.
- The maximum number of parties active on a bridged call is five (no more than two of these may be outside parties).

Incoming calls ring the principal station and its bridging stations according to ring options specified by the System Administrator. The principal station can be administered to send, or not send, ringing to the bridging station; both stations can be administered to receive immediate ring, delayed ring, or no ring for incoming calls. Table 2-F summarizes the effect of different settings for these ring options.

Table 2-F. Bridged Ringing Options

Principal Station		Bridging Station		
		No Ring Administered	**Immediate Ring Administered**	**Delayed Ring Administered**
Administered to send ring on no answer?	yes	no ring	immediate ring	delayed ring
	no	no ring	no ring*	no ring*
Administered to send ring on busy?	yes	no ring	immediate ring	delayed ring
	no	no ring	no ring*	no ring*

* When Send All Calls is enabled at the principal station, immediate ringing occurs in these situations.

As with standard System 25 operation, off-hook stations receive abbreviated alerting instead of repeated ringing. In addition, ring options have no effect on visual alerting via status LEDs on SA or BA buttons.

When an inside or outside call comes into an SA button of an idle principal station, it and each bridging station receive ringing, according to options set by the System Administrator, and visual alerting with a flashing status LED. If one of the users picks up the call, the green

status LEDs of the SA button and the bridged appearances light steadily. If neither principal nor bridging user answers, the call goes to the principal's coverage—if provided—in the usual way. When the covering station answers, the status LEDs at the principal and bridging stations light.

When a call comes into an idle SA button of a principal who is busy on another SA button, the bridging station(s) will receive flashing (and ringing, if administered) on the bridged appearance of the called button. A bridging station can answer the call.

A call to a principal with all SA buttons busy will receive busy tone unless coverage is available or Call Waiting is administered. Bridging stations do not have access to the call.

Considerations

The bridging feature meets the needs of executive/secretary type arrangements where both parties place and receive calls on the same extension numbers. Bridging allows more complete coverage of all incoming calls. It provides options that can enable several call-answering patterns.

It is recommended that each SA button at a principal station have a corresponding BA button at the bridging station. With this arrangement, the bridging user can track all calls coming to the principal's SA buttons.

Except for their bridging functions, bridging stations operate independently of their associated principal stations.

The bridging feature applies only to calls appearing on the SA buttons of a principal station. Calls on Automatic Intercom, Personal Line, DSS, and Flex DSS buttons are not accessible from a BA button.

Interactions

The following features interact with Bridging.

Abbreviated Ringing: When a call arrives on an SA button of the principal, off-hook stations—principal and bridging—that have bridged ringing enabled receive abbreviated ringing. The green status LEDs of the SA button and the BA button associated with the incoming call continue to flash after the abbreviated ring.

Account Code Entry: A station user can use the Optional Account Code Entry feature for incoming or outgoing calls on bridged appearances. If a bridging station user has the Forced Account Code Entry (FACE) feature, an account code must be entered for all applicable outgoing calls on both SA and BA buttons.

Attendant Positions: Direct Trunk Attendant Consoles cannot serve as principal stations. Switched Loop Attendant Consoles cannot serve as either principal or bridging stations.

Automatic Intercom: Calls on Automatic Intercom buttons on the principal station are not accessible from bridged call appearances.

Automatic Route Selection (ARS): When a station user originates a call on a BA button and dials the ARS access code, the call is completed according to the restrictions assigned to that station. The restrictions of the principal and the other bridging stations have no effect.

Callback Queuing: Calls originated on Bridged Access (BA) buttons can be queued. On callback attempts, only the originator will be rung; all other appearances will only flash. Any appearance in the bridging arrangement can be used to drop a queued call, if no other station is off-hook.

If both principal and bridging users are off-hook on a call to a busy facility, only the first one off-hook can queue the call.

Calling Restrictions: If a station goes off-hook on a BA button and dials a number, the call is completed according to the bridging station's restrictions and characteristics, not the principal station's.

Two bridging stations or a bridging station and its principal station can attempt to originate a call on their corresponding SA and BA buttons at the same time. This call is completed according to the calling restrictions of the station that went off-hook first.

Conference: A station user can make conference calls on BA buttons using the normal conference feature operations. When a call is held for conference by pressing the CONFERENCE button, an idle System Access (SA) button or an idle System Access-Originate Only button, if available, is automatically selected by the system for placing the new call. If neither of these button types is idle, the user can manually select a BA button or any other call appearance button on which to place the new call.

While a bridging station or principal is in the process of setting up a conference call, the green status LED of the held call's BA button or SA button has a broken flutter indication. Other bridging or principal stations that are actively bridged to the call have steadily lighted status LEDs; stations that are not active on the call have winking green status LEDs (indicating that the appearance is on hold).

Coverage—Individual and Group: An incoming call is given individual and/or group coverage according to the coverage specified for the principal. Calls appearing on BA buttons at the bridging stations are not extended to the coverage specified for those stations.

Display: Call descriptor " ^ " appears in position 15 of Screen 1 for a call containing more than two active parties; position 16 contains the actual number of bridgers. The number of parties is displayed at each terminal in a bridged call and is updated as the status changes.

Screen 1 | 324 Zimmer,A^4 |

The " ^ " and the number of bridging parties overwrite whatever was in positions 15 and 16 of the current display.

Direct Group* Calling *(DGC): DGC calls arriving on SA buttons at a principal station can receive bridging treatment at a bridging station.

Direct Station Selection (DSS): Calls from DSS or Flex DSS buttons on the principal station are not accessible from bridged appearances.

Exclusion: If a principal or bridging station presses the EXCLUSION button during a call, all other internal stations on the call will be dropped. In addition, Exclusion will prevent any other internal station from bridging onto the call.

Following: Sign-in and sign-out procedures can be performed at the destination station on either an SA button or a BA button. However, since Following calls always arrive on SA buttons, the destination station must have at least one SA button.

Following calls arriving at a principal SA button are accessible at bridged call appearances of that button.

Forwarding: This feature is station-oriented. It can be activated and deactivated for a forwarding principal station only at an SA button on that station. If forwarding is activated at a BA button on a bridging station, it affects calls to that station only.

Hands-Free Answer on Intercom (HFAI): If a station has HFAI activated, internal calls arriving at this station on an SA button will auto-answer. However, calls arriving at this station on a BA button will ring according to the administered ring option and will not auto-answer.

Hold: A principal or bridging station user who is active on a bridged call can hold the call by pressing the HOLD button. If there is still a bridging or principal station active on the call, the green status LEDs of all associated SA/BA buttons remain lighted steadily. If no other bridging or principal station is active on the call, the green status LEDs of all associated SA/BA buttons wink.

Any of the principal or bridging stations can enter the held call, unless Exclusion has been activated or the maximum number of parties are already connected to the conversation.

Last Number Dialed: The Last Number Dialed feature saves numbers called from either SA or BA buttons.

Line Status and I-Use Indication: The meanings of green line status and red l-use indications on BA buttons are consistent with all other System 25 operation.

Message Waiting: Stations with bridged appearances can have Coverage Message Waiting (COVER-MSG) buttons. By using the COVER-MSG button, the bridging user can check and/or change the status of the principal's Message LED. If the bridging station also serves as an Individual Coverage receiver, the same COVER-MSG button can be used for bridging and coverage messaging needs. Use of the COVER-MSG button is identical to current System 25 operation.

Night Service: Directed Night Service calls ring immediately at an available SA button on the Night Service station, regardless of the administered ring option. Bridged appearances of this SA button will flash but not ring, regardless of the administered ring option.

Park: If a station is active on a bridged call appearance and activates Park, the call is parked on the Personal Dial Code (PDC) number of the principal station, not of the bridging station. If the parked call is not answered, it will return on the principal's SA button.

Personal Speed Dialing: Personal Speed Dialing is a station-oriented feature. If a station dials a Personal Speed Dialing code (#20-#39) while off-hook on a BA button, the system will handle this call exactly as if the code was dialed from this station's SA button.

Pickup: Pickup is a station-oriented feature. Thus, calls ringing at a principal SA button can be picked up by members of the principal's Pickup Group; calls ringing at a BA button can be picked up by members of the bridging station's Pickup Group. If a user dials the Group Pickup access code while active on a BA button, the system interprets this as an attempt to pick up a call in the Pickup Group of the bridging station, not of the principal station.

Calls ringing at either a principal SA button or an associated BA button can be picked up by using the Directed Pickup feature.

Pooled Facility—Dial Access: A station originating a call on a BA button and using a facility access code is granted access to that pool according to the Calling Restrictions assigned to the bridging station, not the principal station.

Prime Line Preference: A BA button can be specified as the preferred line for outgoing calls when the station goes off-hook.

Program A bridging station user can program only the Personal Speed Dialing codes, REP DIAL buttons, and FLEX DSS buttons associated with the bridging station, not with the principal station.

Repertory Dialing: If a station user selects a bridged appearance for an outgoing call and then depresses a REP DIAL button, the digits programmed into the button are outpulsed as they would be if the user had selected one of the station's own SA buttons.

Ringing Line Preference: If a station has ringing line preference enabled and has a ringing bridged call appearance, an on-hook user is connected to the bridged appearance if the set goes off-hook. This is the same as current System 25 operation.

Send All Calls: The principal station can be administered so that pressing the SEND ALL CALLS button will send ringing for incoming calls to its coverage stations only, to its bridging stations only, or to both.

If ringing is sent to a BA button via Send All Calls, and if the BA button is administered to not receive ringing, the call will flash (but not ring) at the BA button. If ringing is sent to a BA button via Send All Calls, and if the BA button is administered to receive ringing (immediate or delayed), then the call will ring *immediately* on the BA button.

Station Message Detail Recording (SMDR): When an outside call is answered or originated at a BA button, the SMDR record for this call will report the bridging station's PDC number under the STN column and the principal station's PDC number under the PDC column. If the Call Accountability feature is used when originating a call at a BA button, the PDC column will contain the *accountable* (entered) PDC number in place of the principal's PDC.

If two bridged stations attempt to originate a call at the same time, and if the call is completed, the PDC number of the station that dialed the first digit is placed in the SMDR records under the STN column.

Transfer: A call can be transferred from a bridged call appearance using the usual transfer operations. When a call is held for transfer by pressing the TRANSFER button, an idle System Access button or an idle System Access-Originate Only button, if available, is automatically selected by the system for placing the new call. If neither of these appearance types is idle, the user can manually select a BA button or any other call button on which to place the new call.

The transfer operation and status indications of the principal's SA button (and its corresponding BA buttons) are similar to Personal Line operation with the following exceptions: Calls can be from/to an inside station or from/to an outside location, and after the transfer is completed (transferring station goes on-hook) the call will stay at the principal's SA button and its BA buttons only if one or more of these stations is bridged to the call. Otherwise, the call will be removed. An on-hold bridging station or principal is not considered to be bridged to the call.

Administration Requirements

Voice Terminal Port, Bridging Station:

- Assign a BA button for each SA button on the principal station.
- Assign ringing option for each BA button (no ring, immediate ring, or delayed ring; default = immediate ring).

Voice Terminal Port, Principal Station:

- Assign ringing for each SA button (no ring, immediate ring, or delayed ring; default = immediate ring).
- Send ringing to bridged appearances on no answer? (yes or no; default = yes).
- Send ringing to bridged appearances on busy (off-hook)? (yes or no; default = yes).
- Modify the assignment of the Send All Calls button:

  Abbreviated ring reminder (yes or no).

  Send ringing to bridging stations or coverage stations or both; default = both.

System:

- Search for stations that have bridged call appearances of a principal.

Busy-To-Idle Reminder

Description

Busy-to-Idle Reminder alerts a multiline voice terminal user by a single ring as soon as a busy internal station, DGC group, or facility (trunk group) becomes available. The feature can be activated only for stations, DGC groups, and trunk groups represented on the terminal by DSS, FLEX DSS, AUTO ICOM, and FACILITY buttons.

Before making a call to a station, the multiline voice terminal user can check the green status LED of the station button. If it is lit, the station party is off-hook. To be alerted when the party hangs up and is available again, the user (while on-hook), presses the button of the station. The red l-use LED lights, indicating that Busy-to-Idle Reminder is in effect. When the other party hangs up, the user's terminal rings once. The user simply goes off-hook, and the station is called; the user does not have to press the button again.

If the user calls a station by pressing a FLEX DSS or DSS, button and receives busy tone, the user must hang up before activating Busy-to-Idle Reminder.

When all the trunks in a pool represented by a FACILITY button are busy, the green status LED is lighted. The user can activate Busy-to-Idle Reminder in the same way as for a station call, by pressing the FACILITY button while on-hook. When a trunk becomes idle, the terminal rings once. The user goes off-hook and if the trunk is still available is automatically connected to the trunk. To complete the call the user dials the desired outside number.

Considerations

Busy-to-Idle Reminder gives the multiline voice terminal user a way to get quick access to a station or trunk group that has just become available after being busy. Access to the station or trunk is not reserved for the user who activates this feature; any other user has equal access to the idle facilities.

On some multiline voice terminals, FLEX DSS, DSS, AUTO ICOM, and FACILITY buttons do not have status and l-use LEDs. This makes Busy-to-Idle Reminder less convenient to use because the user must first call the facility to determine if it is available. If it is not, the user activates the feature by hanging up and pressing the button again. When the reminder ring sounds, the user must then remember which button was used to initiate the call.

Interactions

Display: When a user receives this signal, the display format is the same as when the call was originally placed, except that idle descriptor "I" appears in position 1. Number and name fields are displaced to the right.

Busy-to-idle Reminder cannot be used with the Last Number Dialed, Personal Line, or Repertory Dialing features.

Call Accountability

Description

This feature allows calls made by system users from other users' stations to be charged to the callers' own "home" Personal Dial Code (PDC).

To charge a call to the home PDC, the user dials ## followed by the PDC immediately upon receiving first dial tone to place a call. When second dial tone is returned, the user dials the desired number in the normal way,

After completion of the call, the SMDR record will reflect the "accountable" PDC (that is, the caller) in the "PDC" field, and the PDC of the voice terminal used in the "STN" field.

Considerations

Call Accountability, if used consistently, helps to ensure that calling costs are attributed accurately to the personnel who incur the costs. Users do not use this feature when calling from their own stations or when making inside calls from any station.

Interactions

All of the following conditions apply only when a user is calling from another user's station.

Account Code Entry, Forced: The account code entry can be made before or after the Call Accountability entry. Dial tone is returned to the user after either entry.

Account Code Entry, Optional: This feature can be used on the same call with Call Accountability.

Call Accounting System: The caller's PDC that is entered by the Call Accountability procedure is integrated into the reports generated by Call Accounting systems.

Callback Queuing: Any call accountability information entered before activation of queuing is saved for SMDR.

Display: The characters and digits dialed to charge a call are displayed, followed by the called number.

Pooled Facility—Direct Access: When a call is made using a FACILITY button, ##PDC must be dialed before pressing the button.

Remote Access: Remote Access callers cannot use Call Accountability.

Repertory Dialing: When a call is made using a REP DIAL button, ##PDC must be dialed before pressing the button.

Speed Dialing: When a call is made using Speed Dialing, ##PDC must be dialed before dialing the Speed Dialing code.

Call Accounting System (CAS)

Description

Call Accounting is the collecting, processing and use of information about all trunk calls placed from and received by System 25. It is intended to help customers control telephone use and manage associated costs.

Detailed call data is available at Interface Port 2 of the Digital Switch's CPU/MEM Circuit Pack. This data can be fed to a Call Accounting System (CAS) for preparing a variety of cost estimate reports and for providing management and directory type services. For additional information on the call data available at Interface Port 2, refer to the Station Message Detail Recording (SMDR) feature description.

Three station features of System 25 are also related to Call Accounting and are covered in separate feature descriptions. Account Code Entry, both Forced and Optional, allows individual voice terminal users to associate specific account codes with their calls, when necessary. Call Accountability provides users with the means to properly identify calls they make from stations other than their own. The information gathered from these two features is part of the data output from the processor to the CAS.

Two types of CASs can be used with System 25:

- CAS Model 200, 300, 500, or 2000 Software Package associated with an AT&T Personal Computer (PC) 6300.
- CAS Model 200, 300, 500, or 2000 Software Package associated with a Master Controller (UNIX PC).

CAS Models 200, 300, 500, and 2000:

The System 25 SMDR interface provides direct output to the CAS software running on either an AT&T Personal Computer equipped with MS-DOS (V2.11 or later) or a UNIX PC Master Controller. CAS calculates the cost of calls and provides basic and sophisticated call reports.

After a telephone call is completed, System 25 sends a call record to the AT&T Personal Computer via the SMDR interface channel. The PC must be equipped with and running CAS software. Call records are collected by the PC and held in a buffer until they are processed. When a call record is processed, a cost is calculated and assigned to it. That cost, along with other call record information, is then stored on a hard disk for subsequent retrieval.

Two modes of operation are available for PC operation:

- *Dedicated Mode:* The PC is dedicated to one and only one task—processing call records.

- *Multi-Function Mode:* Allows the user to print reports, edit files, and run other PC-based programs while the CAS continues to collect and buffer call records in the background. The user must enter the Dedicated Mode to process calls and generate reports.

The CAS performs three main functions; (1) call record processing, (2) report generation, and (3) CAS system management. In addition, a limited directory lookup and message center is provided. The following is a brief description of each function:

1. *Process Calls:* Involves screening call records, calculating the cost of valid calls, and storing the call records.

2. *Generate Reports:* Allows the user to print the stored call record information organized in one of several different ways. Users can select a report or set Up their own special combination of reports from the following:

   - Summary Reports—A collection of reports that condense and summarize call record information by total number of calls, duration, and cost. The reports can be organized by department, call type, cost center, trunk, extension, cost, duration, time of day, date, and account code.

   - Organization Detail Report—A detailed report of each call record in the system, sorted by department, cost center, and extension.

   - Selection Report—This report can contain at a user's option, summary or detailed information based on any combination of the following items: time of day, date, cost, duration, extension, access code, account code, dialed number, call type, department, or cost center.

   - Account Code Detailed Report—A detailed report on call records sorted by account code. This report can be used for billing clients for calls made in their behalf.

   - Preselected Reports—Allows up to five predefine reports, which can include any of the above mentioned reports. These reports can run upon request or at a specified time and date.

3. *System Management:* Allows the user to perform several functions. These include editing the table of departments, cost centers, and extensions; setting up account codes; defining preselected reports; and keeping call rate information up to date. System configuration may be changed, This allows the user to inform the CAS of changes in System 25 (e.g., dial access codes, trunks) or changes in charge rates. System housekeeping may also be performed. This includes establishing passwords, deleting call records, determining call processing options, and performing various disk maintenance operations.

4. *Directory Lookup and Message Center:* Allows the user to look up anyone by last name, first name, or extension. Messages can be recorded for individuals and can be printed or displayed.

The following table summarizes CAS station and account code capacities.

AT&T CALL ACCOUNTING SYSTEMS		
CAS Model	**Stations**	**Account Codes**
200	100	5000
300	150	5000
500	500	5000
2000	2000	15000

Refer to the CAS documentation supplied with the software package for additional information.

Considerations

The CAS provides customers with an efficient tool to control and manage their telephone usage and costs. The information available can be used to facilitate cost allocation, traffic analysis, and abuse control.

Administration Requirements

System:

- Send SMDR Records To SMDR Port (yes or no; default = yes).
- Minimum length (seconds) of calls that are reported by SMDR (10-255; default = 40).

Hardware Requirements

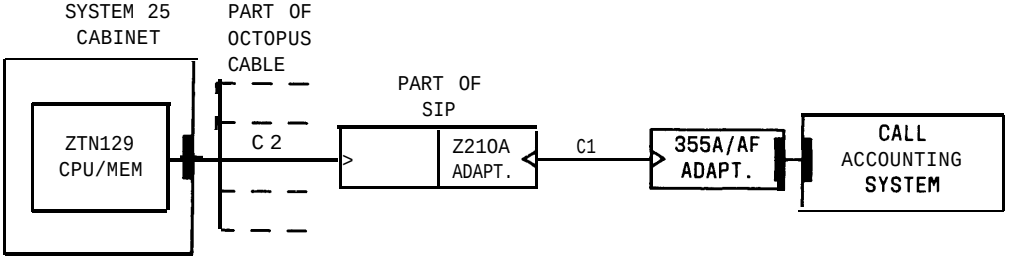
CAS Model 200, 300, 500, or 2000 applications software must be run on an AT&T Personal Computer equipped with MS-DOS (V2.11 or later) or a UNIX PC Master Controller.

SMDR port parameters are as follows:

- No parity; bit is set to zero.
- 1 start bit, 1 stop bit, and 7 data bits.

- Baud rate defaults to 1200 (can be set to 300).
- DTR (data terminal ready) required from printer.
- RTS (ready to send) and CTS (clear to send) not required.
- No flow control.

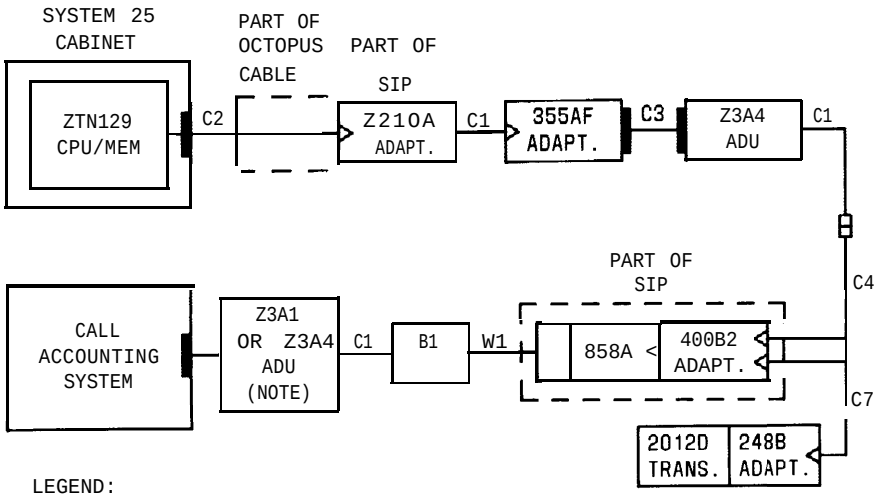
Refer to Figures 2-11 and 2-12 for typical CAS connection information.



LEGEND:

C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
355A ADAPTER - RS-232 PLUG TO MODULAR JACK - PEC 2750-A24
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-11. Call Accounting System—On-Premises Direct Connections (Sharing Same AC Outlet)



LEGEND:

B1 – TYPICAL-103A CONNECTION BLOCK*
C1 – MODULAR CORD (D8W-87) - PEC 2725-07G
C2 – OCTOPUS CABLE (WP90780) - PEC 2720-05P
W1 - 4-PAIR INSIDE WIRING CABLE*
Z3A1 ADU - EQUIPPED WITH 3-FOOT PLUG-ENDED EIA CORD –PEC 2169-001
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD -PEC 2169-004
C3 – EIA CROSSOVER CABLE (M7U-87)
C4 – ADU CROSSOVER CABLE (D8AM-87)
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD
C1 - MODULAR CORDS (2) (D8W-87)
C7 - MODULAR POWER CORD (D6AP-87)
248B ADAPTER - MODULARIZES 2012D TRANSFORMER
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE

} PEC 62515

NOTE: CAS MODEL 200 REQUIRES Z3A4 ADU.

* - FURNISHED BY INSTALLER

Figure 2-12. Call Accounting System—On-Premises Direct Connections (Greater Than 50 Feet From System Cabinet or Nonsharing Same AC Outlet)

Callback Queuing

Description

This feature provides System 25 users with a simple way to complete calls to busy facilities (stations or trunk groups), without having to manually repeat the calling procedures. Callback Queuing puts inside calls to busy stations and trunk pools into a queue. The maximum number of queue slots is 64, administrable in any combination of inside and outside calls. After a call is queued for the busy facility, the caller can stay off-hook or go on-hook. When the queued-for facility becomes free to receive another call, the system directs the longest waiting call to the facility and signals its originator that the call is being completed.

Each station can activate Callback Queuing manually or can be administered for automatic activation of inside calls only, outside calls only, or all calls. Manual activation is by dial access at single-line sets and by operation of the RECALL button at multiline sets. Automatic activation, if administered, occurs whenever a busy facility is called and requires no action by the caller; it can be canceled manually.

Inside Calls

An inside station is considered busy if all its System Access (SA) buttons are in use (multiline sets), if it is off-hook (single-line sets), and if all coverage points are busy. A call to such a station will receive busy or special ringback tone if the calling station is not administered for automatic queuing; the caller can then activate queuing manually. If the calling station has automatic queuing, the caller hears queuing tone (five short beeps) instead of busy or special ringback, and the call goes into queue.

- Manual Queuing

  A single-line terminal user activates manual queuing by flashing the switchhook to get recall dial tone and then dialing the callback access code #60. A multiline user presses RECALL to manually queue for a busy station. Queuing tone is returned if the call can be queued; if not, reorder tone is heard, and the caller must try again later.

- Automatic Queuing

  When the user of a single-line or multiline terminal administered for this option calls a busy station, queuing tone is returned immediately. If maximum queue size has been reached, the caller gets busy tone and must try again later.

- Off-Hook Queuing

  After a call has queued for a busy station, the caller can wait off-hook for the connection to be completed. The caller hears dequeuing tone (three short beeps) when the queued-for station becomes available and then ringback as the station is being rung.

While off-hook on a queued call, a single-line user can transfer or conference the call.

- On-Hook Queuing

  The caller can hang up after a call to a busy station goes into queue. The call retains an association with the on-hook calling terminal, and a callback will alert the on-hook user when the queued-for station is available. After a single-line terminal user goes cm-hook, the queued call cannot be accessed until a callback attempt occurs. During the waiting period, the single-line user can place or receive other calls.

  When a multiline user hangs up after a call is queued, the green status LED of the queued System Access (SA) button winks, as if the call is on hold. If the user goes off-hook and presses the button while the call is still queued, the LED lights steadily and queuing tone is heard again. While waiting for a queued call, the user can place and receive other calls.

- Callback

  Callback applies only to stations that went on-hook after queuing was activated. When the queued-for station becomes available to receive a queued call, the system sends repeated priority ringing to the on-hook calling station. At a multiline set, callback rings at the same SA button where the call to the queued-for station was originally placed; the winking status LED of the button changes to flashing. The on-hook station continues to get priority ringing until the callback is answered or until the administered number of callback rings has been reached. When the original calling station answers the callback, dequeuing tone is heard; this indicates that the queued-for station will now be rung. The calling user hears ringback until the queued-for station answers.

  An off-hook single-line set cannot receive callback until it is on-hook again. If a multiline set is off-hook and active on another call when callback arrives, it receives one cycle of priority ringing.

  Administration sets the maximum number of callback attempts and the number of rings per attempt. Each time a callback attempt for a given call is not answered, it is counted against the assigned number. If the last allowed callback goes unanswered, the system cancels the queued call.

  If a queued-for station becomes busy *during* callback and the calling station answers, queuing tone is heard, not dequeuing; this callback attempt does not count against the administered number. Another special situation exists when a multiline station fails to answer a callback before ringing stops and the queued-for station remains idle during the interval before another attempt. The multiline station can go off-hook between callback attempts, press the queued button, and ring the idle queued-for station.

- Callback Cancellation

  The user can cancel a call that was queued manually or automatically. At a multiline set, if the call is queued off-hook, the user presses the DROP button, then the queued call button. If the call is queued on-hook, the user must go off-hook, press the queued call button to become active on the call, press DROP, and then press the queued call button again.

  At a single-line set, the user must get dial tone, then dial the callback cancel access code #61.

  The attendant at a Switched Loop Attendant Console cancels a queued call by becoming active on the queued call LOOP button and pressing DROP.

Outside Calls

Only trunk pools can be queued for, not individual trunks. Queuing for trunk pools is similar to queuing for inside stations except for the tones received and the methods of placing calls.

To be eligible for queuing when not using ARS, a trunk pool must have all of its members administered to be queuable by dial access users. To make an outside call, the user dials the access code of the pooled facility, then, after second dial tone, the rest of the desired outside number. The dialing must be terminated by pressing # or by waiting for timeout. If all trunks are busy, and the calling station is administered for automatic queuing for outside calls, queuing tone is returned to the caller. (Note that dialing the complete outside number is required, even if all trunks are busy; this ensures that no redialing is necessary on the queued call.)

If automatic queuing is not administered, the caller hears reorder tone but can queue the call by the appropriate manual method. Reorder tone is also returned if the busy trunks are not administered for dial-access queuing or if all the queue slots are in use. In these cases, queuing cannot take place.

Callback Reservations

When busy facilities are called, both queue slots and queued-for stations are “reserved” before they are actually “seized” by the calling stations. Reservation of queue slots applies only to manual queuing, in which the caller has a choice of activating callback queuing, or not, after hearing busy tone. It ensures that callers can elect to queue in the same order that they get busy tone, not in the order that they press RECALL or dial #60. If a caller reserves or seizes the last available queue slot, new callers will not be able to queue.

Reservation of stations means that the system marks stations busy to new calls until all their callbacks have been attempted. Even if an SA button (or a single-line set, which is equivalent to an SA button) is idle, it cannot receive a call while callback attempts remain. However, the station user can place calls.

Considerations

Callback Queuing saves time for users because they can avoid repeated redialing of busy numbers. It allows trunks to be used more efficiently and can reduce the number of trunks required for a system.

The feature is similar to the Busy-to-Idle Reminder feature but applies to different types of calls. The two features can both be used to reduce redialing effort.

A single-line voice terminal can queue only one call at a time.

Calls originated by using Personal Line, Direct Facility Access, and Direct Station Selection buttons cannot be queued.

On calls to busy facilities, Callback Queuing occurs only after Coverage, Following, Forwarding, and Hunting have been attempted.

Interactions

The following features interact with Callback Queuing.

Account Code Entry: An account code entered before queuing is saved for SMDR.

Attendant Camp-On: Trunk calls camped onto a station by an attendant are given priority over queued calls. Multiple camp-on calls are allowed per station.

Attendant Direct Extension Selection: Callbacks to the attendant do not flash at the associated Selector Console.

Attendant Positions: The attendant can queue calls that are extended using the normal START-RELEASE button operation. However, calls originated using only the START button (no other call put on hold) cannot be queued.

Attendant Message Waiting: An attendant active on a queued inside call can toggle the MESSAGE LED of the queued-for station by pressing the ATTENDANT MESSAGE WAITING button.

Automatic Route Selection (ARS): Implementation of Callback Queuing affects ARS in three respects:

- On-hook ARS queuing is allowed.
- Callback Queuing and ARS share a common queue, which has a capacity of 64 calls.
- When a station without automatic queuing originates an ARS call, the caller hears reorder tone if all routes are busy; the call can then be queued manually.

Bridging of System Access Buttons: Calls originated on Bridged Access (BA) buttons can be queued. On callback attempts, only the originator will be rung; all other appearances will only flash. Any appearance in the bridging arrangement can be used to drop a queued call, if no other station is off-hook.

If both principal and bridging users are off-hook on a call to a busy facility, only the first one off-hook can queue the call.

Call Accountability: Any call accountability information entered before activation of queuing is saved for SMDR.

Call Waiting: If a station with automatic queuing calls a busy station with Call Waiting, the calling station hears queuing tone, not special ringback; also, the called party does not hear Call Waiting tone. Call Waiting tone is heard only when special ringback is returned to the caller. A station without automatic queuing gets special ringback but can manually queue the call.

If the queued-for station dials the Call Waiting pickup code *9, the first off-hook queued or waiting call will be dequeued.

Conference: A queued call can be part of a conference, unless a Call Waiting call is already part of the conference. A queued call counts as two conferees until it is completed.

Coverage: Callback calls to the originator do not send ring signals to its coverage stations. If a call is queued for a station, then one of the coverage stations becomes available, the call remains queued for the originally-dialed station.

Data/Pooled Modem: Calls to busy data ports can be queued. Data ports cannot queue calls or receive call waiting treatment. A call requiring a pooled modem cannot queue.

Direct Group Calling (DGC): Inside calls to busy DGC groups can be queued. Queuing is not allowed if all members of the DGC group are logged out. A multiline DGC member with a queued call and a single-line member with an off-hook queued call are considered busy.

Display: Before Callback Queuing is invoked, the display shows the standard format for origination of an inside or outside call. When Callback Queuing goes into effect for the call, the display updates to CALL QUEUED. If the user cancels queuing, the display is QUEUE CANCELED. If the queuing attempt is denied, QUEUE DENIED appears.

When a station receives callback, indicating that the called facility is now available, the display shows the same information seen before queuing, except that queue descriptor Q appears in position 1, displacing the number and name fields to the right. Once the user answers the callback, the display updates to standard origination format; Q is removed.

If a user with Automatic Incoming Call ID is off-hook when a callback attempt is made, the display will not flash the callback call's information. However, this information is accessible via the Inspection feature.

Drop Button: If the user is off-hook on the queued call button, pressing the DROP button and then the queued call button cancels the call.

Exclusion: Pressing the EXCLUSION button does not drop a queued call. The EXCLUSION button's status LED tracks the status LED of the associated button. For example, on a callback attempt, the EXCLUSION LED will also change from winking to flashing. If the EXCLUSION button is tracking a conference on hold, it will stay winking with the rest of the conference. When the callback attempt is answered, the EXCLUSION LED lights steadily to track all the conference buttons.

Following/Forwarding: Calls that forward or follow are queued on the busy "away" station, not the "home" station.

Callback attempts to the originator do not follow or forward.

Hands-Free Answer on Intercom (HFAI): Callback calls to the originator do not receive HFAI treatment.

Hunting: If all stations of a hunt group are busy, the call queues only for the dialed station in the group.

Leave Word Calling: A user who is queued for access to a busy station can invoke Leave Word Calling (LWC). The callback request is canceled when LWC is activated.

Park: Queued calls cannot be parked unless they are part of a conference. Reorder tone is returned whenever an illegal park is attempted, but the queued call is not disconnected. If parked conference members drop out, leaving only a queued call, it will be disconnected to prevent the illegal condition of a single queued call being parked.

Pickup: A callback call cannot be picked up.

Recall/Centrex: The RECALL button can still be used to send switchhook flash to Centrex trunks. If a conference exists with a queuable tone and a Centrex trunk, the first push of RECALL queues a call. A second push of RECALL is needed to send switchhook flash.

Selector Console: Callbacks to the attendant do not flash on the Selector Console.

Send All Calls: Callback attempts to the originator are not affected by Send All Calls.

Transfer: Queued calls can be transferred. Single-line sets can transfer queued calls only before going on-hook. The transferring station must wait for the transferred-to facility to answer before completing the transfer; the transferred-to facility then receives queuing tone. Queued calls cannot be transferred to a tone (ringing, busy, etc.).

Administration Requirements

System:

- Assign the maximum queue size for inside calls (0 to 64; default = 64).
- Assign the maximum queue size for outside calls (0 to 64; default = 64).

Note: It is highly recommended that queue size be set to either 0 or 64, not to some number in between.

- Assign the minimum time between callback attempts for inside calls (0 to 120 seconds; default = 30).
- Assign the minimum time between callback attempts for outside calls (0 to 120 seconds; default = 30).

Voice Terminal Ports: (Every station has independent control of the number of callback retries and the number of rings per attempt.)

- Assign the number of callback retries for inside calls (0 to 15; default = 2).
- Assign the number of callback retries for outside calls (0 to 15; default = 2).
- Assign the number of rings per callback attempt for inside calls (2 to 15; default = 3).
- Assign the number of rings per callback attempt for outside calls (2 to 15; default = 3).
- Allow automatic queuing for inside calls (yes or no; default = no).
- Allow automatic queuing for outside calls (yes or no; default = yes).

Trunk Ports:

- Allow queuing of dial access calls (yes or no; default = no).

Calling Restrictions

Description

Designated voice and data terminals can be restricted from making certain types of calls. Available restrictions are:

- Outward Restriction
- Toll Restriction
- Facility Access Restriction
- ARS Restrictions.

Note: Each of these restrictions is voice terminal oriented, not PDC oriented.

Outward Restriction:

When outward restricted, a station will be unable to place any outside calls. The station will be able to answer incoming calls and place and receive inside calls. A station that is outward restricted will be unable to use Automatic Route Selection to place external calls except to the emergency numbers.

Toll Restriction:

Allows calls by restricted terminals to be made based on as many as the first six digits of the number called (after the facility access code). Toll restricted users can make outgoing calls only to those numbers that are on the Toll Call Allowed (TCA) Lists to which it has access. TCA entries must be in the form NXX or NPA-NXX (exactly three or six characters.) The system administrator can establish up to 4 individual lists. A list can contain from 1 to 64 entries provided that the total of all four lists does not exceed 64. One character "." can be specified as a wild card character in place of the last 1, 2, or 3-digits (e.g., "NX.", "N..", or "...") of the NXX code, but not in the NPA code. When this character is used, any character in the dialed number appearing in that position is acceptable. Those stations assigned Toll Restriction Class 1 have access to all four TCA Lists; Class 2 stations just lists 2 through 4, Class 3 stations just lists 3 and 4, Class 4 stations just list *4.* The entry of an area code followed by "..." on a TCA List allows access to all office codes in that area. To allow calls within a customer's local area, individual office codes are entered; this allows the customer to restrict toll calls within the local calling area. NPA-NXX entries allow specific office codes to be called within an area. Note, NPA *only* entries are not permitted (use NPA-...).

Note that stations that are toll restricted are only toll restricted on CO trunks (type 701 and 801) or when they use the ARS feature. They will not be toll restricted when they dial access (or button access) any other type of trunk (e.g., FX, WATS, or Tie trunks).

Facility Access Restriction:

Any station may be denied dial access to the local CO and/or to all other pooled facilities (as a group). A station so restricted may only *dial* access those facilities via the Automatic Route Selection (ARS) feature.

In addition, each trunk and Virtual Facility can be administered to allow or restrict dial access. If dial access is restricted, the trunk or Virtual Facility may only be *dial* accessed via ARS.

ARS Restriction:

Special restrictions on each station may be imposed when the call is routed by the ARS feature. Facility Restriction Levels (FRLs) are used to restrict access to trunk groups. An FRL is a single digit (0, 1, 2, 3). A terminal assigned an FRL of 0 has the least privileges, a terminal assigned an FRL of 3, the most. An FRL is also assigned to each route in each ARS routing pattern. The terminal's FRL must be equal to or greater than the route's FRL in order to use that facility.

Considerations

Restrictions are used whenever it is necessary to restrict certain users from accessing designated facilities. A typical application is to deny most stations dial access to all trunk groups. This forces callers to use the ARS feature, which should result in reduced toll charges.

Interactions

The following features interact with Calling Restrictions

Automatic Route Selection: The use of the ARS feature will not allow users to avoid restrictions. Outward Restriction and Toll Restriction, when administered, can prevent calls originating at associated voice terminals from routing via ARS. Facility access restrictions, however, are circumvented.

Bridging of System Access Buttons: If a station goes off-hook on a Bridged Access (BA) button and dials a number, the call is completed according to the bridging station's restrictions and characteristics, not those of the principal station's.

Two bridging stations or a bridging station and its principal station can attempt to originate a call on their corresponding System Access (SA) and BA buttons at the same time. This call is completed according to the calling restrictions of the station that went off-hook first.

Callback Queuing: Restrictions in effect at the time a call is originated also apply to the retry attempt.

Forwarding: When Forwarding to an outside station is initiated, the system will ensure that the forwarding does not violate any calling restrictions applied to the forwarding station.

Pooled Facility—Direct Access: Toll restricted stations receive standard toll restriction treatment on all Direct Facility Access (FACILITY) button calls.

Personal Lines: Personal Lines are subject to the toll restriction options of the stations on which they appear.

Remote Access: A barrier code class of restriction (COR) has the same parameters as the class of service permissions associated with stations. A system-wide default COR must be administered for use if barrier codes are disabled. Barrier code CORs override the default COR.

Repertory Dialing: A user cannot use Repertory Dialing to access a number that he/she is restricted from dialing.

Speed Dialing: A user cannot use Speed Dialing to access a number that he/she is restricted from dialing.

Transfer: A non-restricted user (typically the attendant) can transfer a CO trunk to an outward restricted or toll restricted station, giving the station outward service. The toll restriction class of the *transferring* station will apply for calls placed over a transferred trunk.

Administration Requirements

Terminal Port:

- Restrict access to CO trunk pool (yes or no; default = no).
- Restrict access to all other trunk pools (yes or no; default = no).
- Restrict outward calls (yes or no; default = no).
- ARS Facility Restriction Level (Level Number 0-3; default = 3).
- Specify Toll Restriction Class (Class Number 1-4, none; default = none [not restricted]).

Automatic Route Selection:

- Route Facility Restriction Levels.

Trunk Port:

- Allow dial access to this trunk (yes or no; default = yes).

System:

- Allow dial access to this virtual facility (yes or no; default = no).

Call Management System (CMS)

Description

The Call Management System (CMS) is an automatic call distributor that directs specified incoming calls to assigned “agents” for handling. It also provides reports of CMS call traffic and agent performance. The CMS has options and parameters that allow the system to be tailored to the individual needs of specific businesses. The CMS consists of software, a personal computer (PC) with a CMS interface card, voice announcement units (VAUs), and a printer.

In a System 25 with CMS, some incoming trunks are assigned as CMS trunks (lines). The CMS lines are organized into line groups according to the types of calls that are expected to be received. One line group, for example, might carry calls made to a service department number, while another group might be assigned to a sales department.

CMS agents are organized into teams called “splits.” Members of a particular split generally answer one type of call. Each agent split is assigned to answer calls for one or more line groups. Each line group must be assigned a main split and may also be assigned a secondary split (for high traffic period backup). Routing of calls to the secondary split is called “intraflow.”

System 25 provides a new button feature, Agent Status for CMS, that is composed of four button sub-types. One sub-type will be used for the CMS PC. The remaining three are used on CMS agent stations to signal the following operational states:

- Logged Out—CMS station not available to receive CMS calls; this is the state an agent enters, by pressing the LOGGED OUT button, when going off duty.
- Available—Ready to receive CMS calls; entered by pressing the AVAILABLE button.
- After-Call-Work (ACW)—Entered by pressing the ACW button so that the agent can complete work on the latest call and not be interrupted by new CMS calls.

CMS has two modes, Night Service (distinct from System 25 Night Service) and Day Service. When Night Service is active, CMS routes calls to a VAU and disconnects them after the message is finished. When the system is in Day Service, a typical call receives the following treatment:

- CMS looks for an available agent in the main split assigned to the line group.
- If agents are available, the call goes to the agent who has been idle the longest.
- If no agent is available, CMS connects the call to a VAU for a delay message. If an agent becomes available while the message is playing, the call goes immediately to the agent.
- If the “forced delay” option is on during Day Service, each call is connected to a VAU and played the entire message *before* being connected to an agent, even if an agent is available.

- If no agent has become available by the end of the delay message, CMS puts the call into the main split's queue of waiting calls. If available on the System 25, Music-on-Hold will be heard by these callers while they wait.
- As soon as an agent in the main split becomes available, CMS will transfer the call at the front of the queue to the agent.
- If no agent in the main split becomes available, and the call at the front of the queue has waited for a predetermined period, the call will be sent to an available agent in the secondary split (if intraflow has been turned on and the secondary split has been administered).

CMS provides a variety of reports that are available on a daily or cumulative (up to 3 months) basis. The Events Log records up to 200 system events and exception conditions. In addition, call traffic reports can be generated for the following:

- Individual Agents (by day or days)
- Splits (by hour or day)
- Line Groups (by hour or day)
- Line Sub-Groups (by hour or day).

CMS Support Features

Three additional features enhance the capability of CMS operations.

- Transfer-Into-Queue

  Allows anyone on System 25, particularly an attendant, to transfer calls into a line group. It also allows an agent to transfer a call to another line group. It is useful in handling calls made to the wrong line group. Calls transferred in this way receive priority treatment in the new line group.

- Service Monitoring

  Enables a CMS supervisor to monitor an agent's calls (without the knowledge of agent or caller) or to join a call when an agent requests help. Service monitoring is useful in the training of agents. The supervisor's terminal requires a Personal Line button for each CMS line to be monitored.

- Assist

  Allows an agent to send a visual (LED) signal to the CMS supervisor to request assistance. The existing System 25 Station-to-Station Message Waiting feature is used for this purpose. An MSG WAIT button/status LED is required at both terminals in each agent-to-supervisor link.

Considerations

CMS is useful for businesses where particular groups or departments receive special types of calls in high volumes. Members of such groups can be assigned to splits. Call completion time is minimized; with calls going directly to a split, attendant assistance is not required.

CMS has the following maximum capacities:

- 28 lines
- 4 line groups
- 28 lines in one group
- 28 agent positions
- 28 agents in one split.

The CMS supervisor can reassign agents to splits and splits to line groups without interrupting service.

Interactions

Refer to the documentation supplied with the CMS for this information.

Administration Requirements

Refer to the documentation supplied with the CMS for this information.

Hardware Requirements

Refer to the documentation supplied with the CMS for this information.

Call Progress Tones

Call Progress tones provide audible feedback on the status of calls during call setup. These tones are heard through the handset or the headset or the speaker, if Speakerphone or HFAI is activated.

- *Busy Tone:* A slow pulsed tone indicating that all facilities for answering the call are in use.
- *Call Waiting/Camp-On Tone:* A single or double burst of tone sent to a busy terminal to notify the user that a call is waiting. A single tone indicates an inside call; a double tone indicates an outside (trunk) call.
- *Call Waiting Ringback Tone (Special Ringback Tone):* Standard ring back with a short lower-pitched tone added at the end; indicates to the calling party that the called party is busy but has been given Call Waiting tone. Call Waiting ringback is repeated until the call is answered.
- *Confirmation Tone:* Three short tones indicating that the system has accepted the instruction entered.
- *Dequeuing Tone:* Three short tones indicating that the called facility is now available and that the call is being completed.
- *Dial Tone:* A steady tone indicating that dialing or feature activation can begin.
- *Dialing Feedback:* Indicates that a digit has been dialed.
- *Queuing Tone:* Five short tones. Indicates that no facility is currently available to place the call, but that the call has been put into a callback queue and will be completed as soon as a facility becomes available.
- *Reorder Tone:* A fast pulsed tone indicating that all trunks are busy, that a dialing error has occurred, that the terminal is restricted from making this call, or that an account code is required but has not been entered.
- *Ringback Tone, Normal:* The tone heard by the calling party indicating that the called station is ringing; repeated until the call is answered.
- *Ringback Tone, Special: (see* Call Waiting Ringback Tone.)

For additional information, refer to “Tones” in Section 5.

Call Waiting

Description

This feature allows a user at a busy voice terminal to be audibly alerted when another party is calling. A voice terminal is considered busy if all its System Access (SA) buttons are in use (multiline sets), if it is off-hook (single-line sets), and if all coverage points are busy.

With Call Waiting, the user hears a distinctive call waiting tone from the handset one time; the caller hears special ringback tone, repeated. Calls from both inside stations and outside locations (on non-DID trunks) receive call waiting treatment at stations administered for this feature.

The called party who hears call waiting tone has these options:

- Ignore the new call and continue with the current call.
- Terminate the current call, hang up, and answer the new call when it rings.
- Put the current call on hold and answer the new call.

A user at a busy single-line terminal flashes the switchhook to hold the current call. Dialing *9 then connects the user to the new call.

At a multiline voice terminal with all SA buttons busy, the user can answer a waiting call on an idle SA-Originate Only or Bridged Access (BA) button. After putting the current call on hold (with the HOLD button), the user presses the SA-Originate Only or BA button and dials *9 to be connected.

Call waiting tone consists of one beep (high frequency tone) for an inside call or two beeps for an outside call. Special ringback consists of normal ringback with a short separate tone added at the end of each cycle. Special ringback continues until the called party answers.

Considerations

Call Waiting improves the chances of incoming calls to busy terminals being answered.

The caller must remain off-hook for a waiting call to be answered.

The Call Waiting feature also applies to calls extended to busy inside stations by attendants (Camp-On). A busy extended-to party receives call waiting tone, and the caller receives special ringback. The call returns to the console after a predefine interval if it is not answered.

Interactions

The following features interact with Call Waiting.

Attendant Camp-On: Trunk calls camped onto a station by an attendant are given priority over other waiting calls.

Attendant Positions: Calls cannot wait at Direct Trunk Attendant Consoles and Switched Loop Attendant Consoles.

Callback Queuing: If a station with automatic queuing calls a busy station with Call Waiting, the calling station hears queuing tone, not special ringback; furthermore, the called party does not hear Call Waiting tone. Call Waiting tone is heard only when special ringback is returned to the caller. A station without automatic queuing gets special ringback but can manually queue the call.

If the queued-for station dials the Call Waiting pickup code *9, the first off-hook queued or waiting call will be dequeued.

Conference: A call receiving special ringback can be part of a conference, unless a queued call is already part of the conference. A waiting call counts as two conferees until it is completed.

Data Stations: Data ports cannot be assigned Call Waiting.

Direct Inward Dialing (DID) Trunks: Incoming DID trunk calls do not wait at busy stations; they receive busy tone.

Display: If a display station has Automatic Incoming Call Identification active, call waiting tone is accompanied by an incoming call message flashed on the screen. The user cannot inspect the message again, because all buttons are busy with other calls. When the user answers a waiting call, the display updates to standard incoming call format.

Send All Calls: A busy station with Send All Calls activated will receive call waiting tones; the caller will hear special ringback. If the busy station then goes on-hook, single-ring reminder will not be given for that waiting call.

Station Hunting: If all members of a hunt group are busy and the originally-dialed station has Call Waiting, the caller hears special ringback until the station becomes available to answer the call.

Administration Requirements

Call Waiting is assigned on a per-station basis (yes or no; default = no).

Command Mode

This feature allows data terminal users to originate data and voice calls and change (or view) their data port options.

(Refer also to the general description of the system's data features in "Data Services Overview").

Command Mode supports digital data endpoints connected via Asynchronous Data Units (ADUs) to ports on a Data Line CP (TN726). Command Mode also supports, with certain restrictions, users of the STARLAN Interface Circuit Pack, ZTN84.

Command Mode is invoked from a data terminal in the idle (on-hook) mode by:

Terminal Optioned For Autobaud	Terminal Not Optioned For Autobaud
Enter Break followed by Return	Enter Break

The terminal then displays the Command Mode menu:

<Data call> <Voice call> <Options> <Hangup>

Menu items are chosen either by positioning the cursor under the desired item (by typing "space" characters) and entering a Return, or by typing the upper-case character in the menu field (e.g., type "D" to choose data terminal dialing or "O" to move to the options sub-menu). Once a user has entered Command Mode, the terminal is considered off-hook and busy to incoming calls until it returns to the idle mode.

<Data call> or <place Data call>:

Refer to the "Data Terminal Dialing" feature description for information on how to dial from your terminal. Once a data call has been set up, either the "Disconnect/Recall Sequence" (see the Permissible Options, Table 2-G) must be sent or <Hangup> selected from the above menu to terminate the data call and return to idle mode. If a data call is not answered, the caller must disconnect by sending a Break.

<Voice call>:

The data terminal user can originate a call for an on-premises source voice or data terminal to a remote terminal by selecting <Voice call> and dialing the required digits. Refer to the "Third-Party Call Setup" feature description for a complete description of this feature.

<Options>:

If the Command Mode menu item <Options> is chosen, the terminal displays the data port's administered options. System default values for each option are also shown. The display shown below is similar to what is actually presented on the screen. (See Tables 2-G and 2-H) for additional information on options.)

OPTIONS	CURRENT	DEFAULT
Speed (highest)	9600	19200 (Auto)
Parity	Even	Even
Mismatch	Yes	No
Local Echo	Yes	Yes
Answer Text	Yes	Yes
Connect Indication	Yes	Yes
Recall Sequence	Br-Br	Br-Br
<exit>	**<Change options>**	**<View options>**

The System Administrator can change options or can authorize data terminal users to change their own options. Selection of <Change options> from the sub-menu shown above allows the data terminal user to change the values in the CURRENT column. Refer to the "User Changeable Options" feature description for complete information on this feature.

<Hangup>:

The <Hangup> option can be used to terminate a connection to the data port. This option is needed for AT&T STAR LAN NETWORK endpoints to disconnect from a STARLAN Interface CP through Command Mode. Accessing <Hangup> provides the user with these options:

- <eXit>— Do not hang up. Return to the top level of Command Mode.
- <All> —Hang up (disconnected).

Table 2-G. Partial List of Permissible Data Port (TN726) Options

OPTION	DEFINITION
Speed (61 -68)†	Autobaud, Low *, 300, 1200, 2400, 4800, 9600, 19200
Parity (69)	Odd, Even, 0, 1, The 0 and 1 choices are not shown on the user's display.
Enable Command Mode (70)	Yes or No. Must be On for Command Mode (i.e., Command Mode Menu display). Not shown on user's options display.
Allow user to change data port options? (71)	Yes or No. Used to enable/disable User Changeable options feature. Not shown on user's options display.
Permit Mismatch (72)	Yes or No. Allows two data endpoints to communicate at different rates.
Local Echo (73)	Yes or No. Determines whether characters from the data equipment will be echoed by System 25 during Command Mode.
Answer Text (75)	Yes or No. Enables call progress messages to be displayed at the *called* data endpoint.
Connected Indication (77)	Yes or No. Yes indicates that users who have Command Mode enabled will receive the "CONNECTED" message when a connection has been established. If Command Mode is disabled, the Data Line port control lead will be "raised" when a connection is established. Usually set to "No" for host computer endpoints.
Disconnect/Recall Sequence (74)	One Long Break or Two Short Breaks; the sequence used to disconnect a data call.

* A terminal whose baud rate is low cannot use the Command Mode feature. Call origination at this terminal must be via Transfer To Data.

† Numbers in ( ) indicate the action numbers used to administer data port options.

Table 2-H. Typical Option Profiles for Data Port Endpoints

Options	Data Term. or PC	Host Computer	Modem (users)	Modem (computer)
Speed (highest)	19200(Auto)	9600 *	Modem Speeds	Modem Speeds
Parity	Even	Even	Even	Even
Enable Command Mode	Yes	Yes	Yes	Yes
Permit Mismatch	Yes	No	Yes	No
Local Echo	Yes	No	Yes	No
Answer Text	Yes	No	Yes	No
Connected Indication	Yes	No	Yes	No
Disconnect/Recall Sequence	Br-Br	Br-Br	Br-Br	Br-Br

* or 9600(Auto) if **not** not used primarily for Host-Host communications

Communications Access Manager (CAM)

The Communications Access Manager (CAM) program facilitates communication between workstations on the AT&T STARLAN NETWORK (STAR LAN NETWORK) and workstations on System 25.

Since CAM has a built-in interface to System 25's Command Mode, it is an ideal communications application for PCs connected to System 25. Detailed procedures for using CAM can be found in the CAM User Guide. The material here provides a brief overview of CAM capabilities.

CAM is an MS-DOS application program that provides an enhanced calling interface and terminal emulation for PCs. CAM, combined with System 25's Third-Party Call Setup feature, provides the capabilities of an integrated voice /data workstation, specifically:

- A 200-entry directory for automatic dialing of voice and data calls
- VT100 terminal emulation with:
  - — file transfer with error checking
  - — unattended remote access operation with mail
- On-line HELP that is accessible from almost anywhere within the program.

CAM runs on the AT&T PC6300 or compatible PC with at least 384K bytes of memory, running MS-DOS Version 2.0 or later (when connected to System 25) or MS-DOS Version 3.1 or later (when connected to the STARLAN NETWORK).

The PC running CAM can be connected to System 25 in one of two ways (Figure 2-13):

1. By the PC's RS-232 COM port to the System 25 via an ADU/DLC connection
2. As a STARLAN NETWORK workstation to the System 25 via the STARLAN Interface CP.

CAM interfaces with System 25's Command Mode to provide call control. The Third-Party Call Setup feature provides voice call origination.

The STARLAN NETWORK communication driver (NAUCOM) is used before CAM is run on a STARLAN NETWORK workstation. The Extended Device driver (CAM232) is used when CAM is run on a PC connected to a System 25 DLC port.

The default screen presented when the user accesses CAM is the phone directory screen. The phone screen is divided into five partitions:

- Call Appearance area—provides call appearance for voice lines and data lines for each extension shown. A call timer for each line is also displayed.
- Feature Selection area—allows the user to select the voice or data line to be used, initiate the call, and start the timer by function keys. Additional function keys may be assigned to Repertory Dialing numbers.

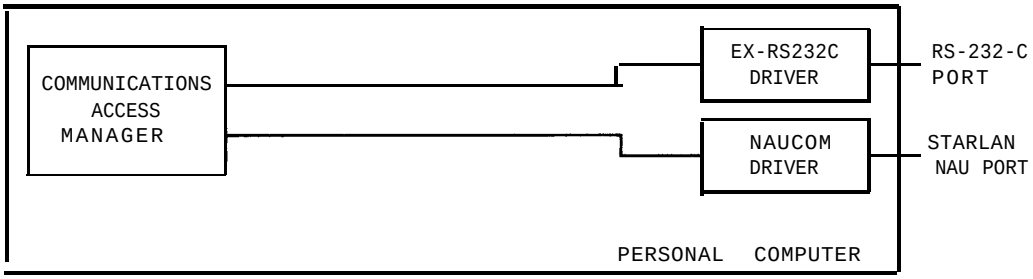


Figure 2-13. Communications Access Manager Architecture

- Personal Directory area—holds a maximum of 200 entries, displayed 10 entries at a time. Each screen is arranged alphabetically.
- Message and Status area—contains prompts and messages for the user for the action being executed.
- Command Line area—contains commands available to the user for the area being worked in. Commands are executed when the user presses the <ALT> key and the first letter of the command.

  The user may access the following commands:

  — Data mode—provides the user with the terminal emulation screen.

  — Edit—provides the user with the directory edit screen. Allows the user to add, modify, and erase directory entries, group names, and feature functions.

  Directory entries contain name, number with auto login script, comment, group, and voice/data fields. Data entries also have parameter setup, a screen with fields for speed, parity, permit mismatch, and number of bits. The parameter setup allows speeds of 2400, 4800, 9600, and Autobaud.

  — Find—allows the user to search directory entries by name or group ID.

  — Restore—displays the first 10 entries of the directory after a Find.

  — Print—prints the entire contents of the directory on device LPT1.

  — Setup—provides the user with the setup screen. Allows the user to view or change the following options: communications port, printer port, speed, parity, character size, return key code, autotimer, flow control, extension numbers, remote access enable, remote access password, and remote greeting.

Conference

Description

This feature allows up to five parties, including the conference originator, to participate in a conference call. Any voice terminal user, including operators at Direct Trunk Attendant Consoles and Switched Loop Attendant Consoles, can set up conferences. Refer to the description of “Conference Drop” for additional information on conferencing.

Multiline Voice Terminals:

Multiline voice terminal users can add another (external or internal) party to an existing call by pressing the CONFERENCE button. This places the first party on Special Hold (indicated by a broken flutter on the line appearance button) and the system selects an idle SYSTEM ACCESS or LOOP (Switched Loop Console) button and provides system dial tone. The user may dial the desired number or select another facility to dial the party to be conferenced-in. Subsequently, pressing the held line button completes the conference. If the facility to be added is busy, the conference will be denied.

Users can conference up to two outside facilities (trunks), and up to five parties in all. Any attempt to add a sixth party will be denied, and the sixth party will be dropped. This limit is for the conference as a whole. Other conference inside stations are also prohibited from adding a third outside party or sixth party.

Single-Line Voice Terminals:

The single-line voice terminal user can establish a conference by momentarily pressing the switchhook, which puts the first party on hold, receiving Recall Dial Tone, and then dialing a second party. After connection to the second party, another press of the switchhook establishes the conference. A third press of the switchhook will drop the second party, restoring the original call. The user cannot put a conference that he/she has established on hold. Other internal conferees (multiline or single-line) may then add additional parties to the conference up to the five party/two outside line maximum.

Considerations

The Conference feature allows any voice terminal user to set up conference calls. Non-attendant users do not need the assistance of the attendant.

Waiting for an added party to answer and announcing the purpose of the call before adding the party to the conference is good operating practice.

A ringing line can be added to a conference and counts as one of the conferees. A queued or Call Waiting call can be added to a conference and counts as two conferees until it is completed; when completed it counts as one conferee.

Interactions

The following features interact with Conference.

Account Code Entry, Forced (FACE): Calls can be conference in both directions between a FACE-restricted station and a non-FACE station.

Account Code Entry, Optional: If more than one user attempts to associate an account code with a Conference Call, the first to activate the feature will prevail.

Attendant Message Waiting: Pressing the Attendant Message Waiting (ATT MSG) button while on a conference call will be ignored.

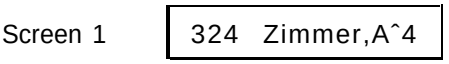
Bridging of System Access Buttons: A station user can make conference calls on Bridged Access (BA) buttons using the normal conference feature operations. When a call is held for conference by pressing the CONFERENCE button, an idle System Access (SA) button or an idle SA-Originate Only button, if available, is automatically selected by the system for placing the new call. If neither of these button types is idle, the user can manually select a BA button or any other call appearance button on which to place the new call.

While a bridging station or principal is in the process of setting up a conference call, the green status LED of the held call's BA button or SA button has a broken flutter indication. Other bridging or principal stations that are actively bridged to the call have steadily lighted green status LEDs; stations that are not active on the call have winking status LEDs (indicating that the appearance is on hold).

Callback Queuing: A queued call can be part of a conference, unless a Call Waiting call is already part of the conference. A queued call counts as two conferees until it is completed.

Call Waiting: A call receiving special ringback can be part of a conference, unless a queued call is already part of the conference. A waiting call counts as two conferees until it is completed.

Display: Call descriptor '' ^ '' appears in position 15 of Screen 1 for a call containing more than two active parties; position 16 contains the actual number of conferees. The number of conferees is displayed at each terminal in a conference call and is updated as the status changes.



The " ^ "" and the number of conferees overwrite whatever was in positions 15 and 16 of the current display.

When a queued call is added to a conference, the associated displays are modified in only one respect: the Q symbol appears as the first character of the queued call display. When the queued facility becomes available and the call is made, "Q" is removed.

When a nondisplay station originates a trunk call, then conferences the call with an inside display station and drops off, the display shows the trunk name only, not the originally-dialed digits.

Exclusion: Exclusion may be invoked before establishing a conference. If it is invoked after the conference is established, all internal conferees will be dropped (except for the party that invoked Exclusion).

Extended Stations: An Extended Station counts as one of the two allowable outside lines on a conference call.

Forwarding: If one of the called parties for a conference is a forwarding station, its forwarded-to station will be the conference facility.

If a conference call is transferred to a forwarding station, it will be given normal Forwarding treatment.

Music-On-Hokf: Music-On-Hold may be enabled or disabled for "Special Hold" through a System Administration item. However, if the outside line is already part of a conference, music is not heard.

Off-Premises Stations (OPS): For conference purposes, an OPS counts as one of the two allowable outside lines.

Paging System Access: A paging zone may not be conference.

Park: Park may be used to place a conference on hold. Parked conference calls do not return to the parking station (they remain parked).

If a 5-person conference is parked, the conferee who parked the conference will be dropped when someone picks up the parked conference.

Remote Access: Remote Access callers cannot use the Conference feature.

Trunk-To-Trunk Transfer: Trunk-to-trunk transfers may be set up using the Conference feature. The conference must include an incoming trunk call on either a ground start, loop start (if trunk-to-trunk transfer is allowed by System Administration), DID, or tie trunk if it is to continue after all inside stations have dropped off.

Conference Drop

Description

This feature allows a voice terminal user, except for the attendant at a Switched Loop Attendant Console (SLAC), to selectively drop a previously added party from a conference call. At a SLAC, the attendant can drop conferees only before they have been added to conference.

Multiline Voice Terminals (except SLAC):

On a multiline voice terminal, pressing the DROP button and then pressing the button appearance of a conference party drops that party from the conference.

If a station called for a conference does not answer, the conferencing user should drop the call by pressing and releasing the switchhook before returning to the conference. Otherwise, the ringing line will be added to the conference.

Switched Loop Attendant Consoles:

Once a conference has been set up and all the parties can talk to each other, the SLAC attendant cannot selectively drop a conferee. Individual members of the conference wishing to drop out must hang up. However, while still setting up a conference, the attendant can drop calls before they have been conference in, as follows:

- A call to an *inside* party rings unanswered or returns busy tone—hang up.
- A call to an *outside* party rings unanswered or returns busy tone—press another LOOP button or RELEASE or FORCED RELEASE.
- A call to an *inside* or *outside* party is completed but the person cannot participate—press another LOOP button or RELEASE or FORCED RELEASE.

It is good operating practice to wait for the called party to answer before adding the party to a conference.

All Multiline Terminals:

If all System 25 stations hang up on a conference with two outside lines, the outside parties will remain conference (until one of them hangs up) if at least one is an inbound call on a ground start, tie, or DID trunk or an inbound call on a loop start trunk if loop start trunk-to-trunk transfer is allowed by System Administration. If not, the call will be terminated when the last inside user disconnects from the conference.

Single-Line Voice Terminals:

A single-line terminal user, after having established a three-party conference, can drop the second party and retain the first party by pressing the switchhook.

Considerations

Conference Drop allows users to conference lines appearing on their terminals and then remove them from the conference when appropriate.

The only parties that a user should try to drop from a conference are those that the user actually added. If a user tries to drop a party who previously added the user to a conference, other parties may also be dropped.

Interactions

The following feature interacts with Conference Drop.

Callback Queuing: If the user is off-hook on the queued call button, pressing the DROP button and then the queued call button cancels the queued call.

Coverage, Group

Description

This feature allows calls to covered stations to be redirected to a group of covering stations.

A total of 32 standard Coverage Groups may be assigned; an additional 32 “DGC” Coverage Groups may be assigned. Each standard group may be covered by up to eight coverage receivers (buttons). There is no limit on the number of covered users (senders) that each Coverage Group can include, but a covered user can be assigned to only one Coverage Group. Each coverage receiver must have a multiline set equipped with a Cover (COVER-GRP) button, except as noted below. A covering set may be assigned more than one COVER-GRP button for the same or different groups.

In systems equipped with a Switched Loop Attendant Consoles (SLAC), the console queue can serve as a standard coverage group receiver. The consoles cannot have COVER-GRP buttons, so the queue directs coverage calls to LOOP buttons.

Direct Group Calling (DGC) Groups may be designated as Coverage Group receivers. This provides the capability for System 25 to support “non-integrated” voice mail systems as well as allow the formation of coverage pools.

Senders may be either single-line or multiline voice terminals. Receivers may be single-line voice terminals only if part of a DGC Coverage Group. Multiline voice terminals may always be used as receivers

Standard Group Coverage

When a call arrives at a voice terminal that has group coverage, the COVER-GRP or LOOP button status LED at the covering voice terminals will flash. Covering voice terminals will begin to ring after a specified number of rings at idle covered voice terminals. If there is no idle system access button at the covered station (sender), the call receives coverage treatment, and the call immediately rings at the covering terminal(s). If no idle cover button is available at the covering terminal(s), the calling party receives a busy signal.

Ringing may be turned off at standard receiver stations for each covering button, as desired (not recommended). If this option is selected, a flashing status LED will be the only indication received at the covering station. In addition, Coverage ringing may be turned off on internal calls (if desired) on a system-wide basis.

A member of a standard receiver group can use the Line Selection (Preelection) feature to answer covered calls even before any audible alerting has begun at the covering user’s terminal. This is useful if the user knows that the covered party is unavailable.

A covered voice terminal may elect to have calls covered while it is busy on another call. Calls directed to an idle button on a busy covered multiline voice terminal will start ringing at the covering terminal after a single burst of ringing at the busy covered voice terminal. If there is no idle Cover button on the covering voice terminal, the system will periodically check for an idle Cover button and ring at the first available coverage receiver. Calls directed to a busy single-line voice terminal will start ringing immediately at the covering

terminal. If there is no idle Cover button on any covering terminal, either individual or group, the call will not receive coverage and the calling party will receive Busy Tone.

Calls sent to coverage will continue to ring at single-line sender terminals but will cease ringing at multiline sender terminals. In the latter case, the calls will remain on the incoming call appearance button, and that button's status LED continues to reflect the status of the call. In particular, covered calls to multiline sets remain available and can be entered by the called (sender) station.

If a covered station activates the Send All Calls feature, calls will be directed to coverage immediately, with or without a single-ring reminder, as administered.

A station can provide (or receive) Individual Coverage (see Coverage, Individual) and also be a member of a Coverage Group (sender or receiver). Unanswered calls to a station, provided both Individual and Group Coverage, will first ring at the Individual Coverage station and then, after a second delay cycle and still unanswered, will ring at the Group Coverage station.

Calls from a covering station to a covered station will not be covered unless the covered station has additional coverage. This is an important consideration when the attendant provides coverage.

DGC Group Coverage

Calls proceeding to the DGC Coverage Receiver Group hunt in a circular fashion for the first idle station, starting from the last station to receive a call. if all DGC members are busy, both internal and external calls continue to ring and/or flash at the covered station and any individual coverage receiver's station(s) until a DGC station becomes idle. If a DGC group is used for both DGC calls and group coverage, trunk calls into a DGC group have priority over coverage calls. Calls sent by coverage to a DGC Coverage Group member station do not receive additional coverage.

DGC groups cannot be coverage senders to another DGC Coverage Group. However, calls made directly to a DGC member can be covered by another DGC Coverage Group.

Once a call has been redirected to a DGC Coverage Group member, the call is transferred to the covering station. The call continues ringing until answered or dropped. The call is not accessible at the covered station nor any individual coverage receiver once it is redirected to an idle DGC station. If all DGC members are busy, the call remains accessible at the covered station until a member is available.

DGC Coverage Groups count against the system specified maximum number of DGC groups, but *not* against the number of Coverage Groups. The limit of eight receiving stations per Coverage Group does not apply when administering a DGC group as a Coverage Group. A maximum of twenty stations per DGC Coverage Receiver Group is allowed.

Considerations

Coverage provides a way to redirect calls to alternate answering positions. The feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special functions, such as the Send All Calls feature, accommodate the day-to-day variations that occur in an employee's work schedule.

Interactions (Standard Group Coverage)

The following features interact with Standard Group Coverage.

Attendant Console, Direct Trunk: If the Direct Trunk attendant is a receiver for a Coverage Group and extends a call (using the Start button or Selector Console) that is unanswered/busy to a member of the group, the call will return on the Return-On-Don't-Answer (RTN-DA) or Return-On-Busy (RTN-BUSY) button, not on the attendant's COVER-GRP button.

Automatic Intercom: Auto-Intercom calls do not receive coverage.

Bridging of System Access Buttons: An incoming call is given group coverage according to the coverage specified for the principal. Calls appearing on Bridged Access buttons are not extended to the coverage specified for those bridging stations.

Callback Queuing: Callback calls to the originator do not send ring signals to its coverage stations. If a call is queued for a station, then one of the coverage stations becomes available, the call remains queued for the originally-dialed station.

Coverage, Individual: Unanswered calls to a station, provided both Individual and Group Coverage, will first ring at the Individual Coverage station and then, after a second delay cycle and still unanswered, will ring at the Group Coverage station.

Coverage/Station Hunting: A call to a busy single-line voice terminal that is both a member of a Station Hunting group and a coverage sender will first hunt for an idle station to service the call. If none is available, the call will be sent to coverage.

Direct Group Calling (DGC): A call to a DGC group member will receive coverage if the member is also a Coverage sender. Calls to a busy DGC group do not receive coverage. Instead, after a predefine number of rings, the call will be transferred to delay announcement (if provided), or ringing will be transferred to all button appearances of the line.

Exclusion: If a coverage receiver invokes Exclusion after answering a coverage call, all other terminals (including the attendant and the covered station) are excluded from the call. The covered user cannot enter the call until EXCLUSION is pressed a second time by the covering user.

Forwarding: When a station has both Coverage and Forwarding in effect, calls are routed first to the forwarded-to station. If not answered there within an administered number of rings, calls ring at the forwarding and coverage stations and stop ringing at the forwarded-to station.

When forwarding to an outside number, coverage may only occur in one case: the forwarding had been activated using a trunk group's facility access code (not ARS), the forwarding cannot be completed because the trunk group is busy, and the forwarding station is not busy. In this case, the call will ring at the forwarding station and its coverage stations.

Hands-Free-Answer On Intercom: An incoming (inside) call will not receive coverage if auto-answer is activated, since the set will answer the call (whether the user is present or not).

Leave Word Calling (LWC): A multiline voice terminal user can activate LWC for the called party even if the call has gone to coverage.

Night Service: Directed Night Service calls do not receive coverage.

Personal Dial Codes: Calls directed to a station because another *non-floating* PDC is signed in there *do not* receive the coverage treatment of the signed-in station. Such calls return to their home station and receive that station's coverage (immediately upon return). Calls to signed-in *floating* PDCs, on the other hand, receive *the same* coverage treatment as any other calls to the signed-into station. They, of course, have no home station to return to.

Personal Lines: Personal line calls receive the coverage of the principal (owner) station for that line. Other line appearances (even if administered to ring) will not receive coverage.

Pickup: Pickup is independent of coverage. When a call is answered via Pickup, all Cover buttons associated with the called party go idle.

Tie Trunks: Tie Trunk calls directed at a user with coverage receive normal coverage treatment.

Transfer: When a covering station transfers a covered call to another station, the call will no longer appear at the covering station's Cover button or at the covered multiline station.

Interactions (DGC Group Coverage)

The following features interact with DGC Group Coverage.

Attendant Console, Switched Loop: If a SLAC is a member of a DGC Coverage Receiver Group, any DGC Group Coverage call sent to this attendant will enter the common queue and be treated as a *coverage call,* not as an Attendant—DGC call. Thus, the call will be handled by whichever attendant is administered to receive coverage calls.

Attendant Direct Extension Selector Console: The Selector Console can be used to transfer and place calls to a DGC Coverage Group provided the DGC group access code appears on the console. The status LED of the DXS button lights steadily whenever all stations in the DGC Coverage Group are busy.

Automatic Intercom: Auto Intercom calls do not receive coverage.

Bridging of System Access Buttons: An incoming call is given group coverage according to the coverage specified for the principal. Calls appearing on Bridged Access buttons are not extended to the coverage specified for those bridging stations.

Callback Queuing: Callback calls to the originator do not send ring signals to its coverage stations. If a call is queued for a station, then one of the coverage stations becomes available, the call remains queued for the originally-dialed station.

Direct Station Selection (DSS): A DSS button can be assigned to a DGC Coverage Group. The button lights whenever all DGC members are busy.

Flex DSS: A Flex DSS button can be assigned to a DGC Coverage Group.

Forwarding: When a station has both DGC Coverage and Forwarding in effect, calls are routed first to the forwarded-to station. If not answered there within an administered number of rings, calls ring at the forwarding station and DGC coverage group and stop ringing at the forwarded-to station. Once the call is directed to a DGC group member, the call appearance is removed from the forwarding station.

When forwarding to an outside number, coverage may only occur in one case: the forwarding had been activated using a trunk group's facility access code (not ARS), the forwarding cannot be completed because the trunk group is busy, and the forwarding station is not busy. In this case, the call will ring at the forwarding station and its DGC coverage group. Once the call is directed to a DGC group member, the call appearance is removed from the forwarding station.

Leave Word Calling (LWC): A multiline voice terminal user can activate LWC for the called party even if the call has gone to coverage.

Night Service: Directed Night Service calls do not receive coverage.

Personal Line Access: All outside lines directed to a DGC group can be assigned to button appearances in addition to the DGC assignment. If the outside lines appear at stations that also have DGC coverage by the same group, then the operation is as follows:

> When an incoming call is ringing in the DGC group, the status LEDs on the appearance buttons light steadily, indicating that the line is busy. If the call goes unanswered after a system-specified number of rings, then a delay announcement is provided. The caller is subsequently put on hold and receives music if available. If the system is not equipped with a delay announcement, the call begins to ring at all line appearances after the system-specified number of rings.
>
> If the outside lines are not directed to a DGC group, but are provided DGC Group Coverage, the feature operation is the same as for incoming calls on SA keys except that the call appearance remains accessible at the covered station after being directed to a DGC Coverage Group member.

Pickup: A DGC Coverage Group member can also be in a Pickup group.

Station Hunting: Calls directed to a DGC Coverage Group will not hunt.

Trunk Groups: Trunks can be directly assigned to DGC groups that also act as coverage group receivers. Among tie trunks, only automatic incoming tie trunks can be translated as directed to a DGC Group.

Administration Requirements (Standard Group Coverage)

System:

- Provide Coverage ringing on internal calls (yes or no; default = yes).
  If “no”, covered calls will flash but not ring at covering stations on internal calls.

- Number of rings before Coverage ringing starts on no answer (0-31; default = 2).

  The status LEDs on Group Coverage buttons at covering stations begin flashing immediately in ail cases. Ringing, in addition to flashing, is always sent on external calls, though it may not be accepted at the covering stations.

Voice Terminal Port:

- Coverage Sender group number (1-32; default = 1).

- Provide Coverage ringing on no answer (yes or no; default = yes).
  If “no”, flashing LED is the only indication received at the covering station; the calling party always hears ringing.

- Provide Coverage ringing on busy (yes or no; default = yes).
  If “no”, flashing LED is the only indication received at the covering station.

  The calling party always hears ringing.

- Coverage Receiver button

  — Group Number (1-32)

  — Allow Ring (yes or no; default = yes).
  If “no”, flashing LED is only indication received at this covering station.

Administration Requirements (DGC Group Coverage)

The parameter to assign a coverage sender group has been expanded to include DGC groups as coverage group receivers. DGC receiver groups are first set up as regular DGC groups, numbered 1-32. DGC *coverage* groups can then be specified by using coverage group numbers 101-132, where coverage group 101 has DGC group 1 as its receiver group, coverage group 102 has DGC group 2 as its receiver group, etc. No button assignments are required.

Send ringing options (on busy, on no answer, system-wide for internal calls) have no effect for DGC group coverage; *ringing is sent for all calls that go to coverage.*

The system search of group coverage sender stations has been expanded to allow the system administrator to enter a DGC Coverage Group and list all its sender stations by PDC.

Coverage, Individual

Description

This feature is very similar to Group Coverage, covered in the preceding feature description. The primary difference is that Individual Coverage is a one-on-one type coverage between pairs of stations.

An Individual Coverage (COVER-IND) button can be assigned on multiline voice terminals to cover calls to a specific (single) voice terminal. The covering station can answer covered calls by selecting COVER-I ND. Each COVER-IND button can be programmed to ring or not ring. If ringing is selected, the covering station will begin ringing after a specified number of rings at the covered station. When the specified number of rings has occurred, covered multiline voice terminals will stop ringing. Covered single-line voice terminals continue to ring until the call is answered at a covering terminal. When the call is answered at the covering station, the call remains accessible at the call appearance button of multiline voice terminals, but is no longer accessible at single-line voice terminals.

Each COVER-IND button at a covering station represents one covered voice terminal. If more than one voice terminal is to be covered, multiple buttons are required, one for each station covered. A covering voice terminal may be assigned multiple COVER-IND buttons for a particular station to cover multiple simultaneous calls to that station. The first button will track the first call, the second button, the second call, etc.

Up to eight COVER-IND buttons can be assigned for each covered station.

There is no limitation on the number of stations that can receive Individual Coverage.

A voice terminal can receive both Individual Coverage and Group Coverage.

Refer to Messaging Services for a description of Coverage Message Waiting service, which allows the covering station to control the status of the covered user's Message LED.

Calls from a covering station to a covered station will not be covered unless the covered station has additional coverage. This is an important consideration when the attendant provides coverage.

Considerations

Coverage provides a way to redirect calls to alternate answering positions. The feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special functions, such as the Send All Calls feature, accommodate the day-to-day variations that occur in an employee's work schedule.

The Individual Coverage feature is not administrable on the Switched Loop Attendant Console.

Interactions

The following features interact with Individual Coverage.

Attendant Console, Direct Trunk: When a coverage call rings at a busy Attendant Console, the attendant receives a single burst of ringing. If the call is still unanswered when the attendant hangs up the other call, the Attendant Console will resume ringing.

Bridging of System Access Buttons: An incoming call is given individual coverage according to the coverage specified for the principal. Calls appearing on Bridged Access buttons are not extended to the coverage specified for those bridging stations.

Callback Queuing: Callback calls to the originator do not send ring signals to its coverage station. If a call is queued for a station, then the coverage station becomes available, the call remains queued for the originally-dialed station.

Coverage, Group: Unanswered calls to a station, provided both Individual and Group Coverage, will first ring at the Individual Coverage station and then, after a second delay cycle and still unanswered, will ring at the Group Coverage station.

Direct Station Selection (DSS): Calls placed via a DSS button to a user with Individual Coverage will receive coverage. When a DSS button is used to activate the busy-to-idle reminder for the user, the reminder is returned only when the user becomes idle, not when an associated coverage user becomes idle.

Exclusion: If a covering station answers a coverage call and then invokes Exclusion, all other inside stations, including the covered one, are excluded from the call.

Forwarding: When a station has both Coverage and Forwarding in effect, calls are routed first to the forwarded-to station. If not answered there within an administered number of rings, calls ring at the forwarding and coverage stations and stop ringing at the forwarded-to station.

When forwarding to an outside number, coverage may only occur in one case: the forwarding had been activated using a trunk group's facility access code (not ARS), the forwarding cannot be completed because the trunk group is busy, and the forwarding station is not busy. In this case, the call will ring at the forwarding station and its coverage stations.

Hold: May be used by the covering user to place a coverage call on hold. The COVER-IND button's status LED winks at the covering station. At the covered station, if the call is on a Personal Line button, the button's status LED winks; if the call is on an SA button, the status LED lights steadily. The held call will automatically leave the coverage terminal if picked up by the covered user. The covering station will be unable to reenter the call.

Leave Word Calling (LWC): A multiline voice terminal user can activate LWC for the called party even if the call has gone to coverage.

Transfer: When a covering station transfers a covered call to another station, the call will no longer appear at the covering station's Cover button or at the covered multiline station.

Administration Requirements

Voice Terminal Port:

- Individual Coverage button (button function # 12):
  - — Individual Coverage PDC (1-9999).
  - — Allow Ring (yes or no).

Data Call Setup

This feature allows a user to originate calls from a data terminal or a voice terminal. System 25 provides three methods of data calling:

- Dialing from a data terminal, which is described in the “Data Terminal Dialing” feature description.
- Setting up data calls from a voice terminal, which is described in the “Transfer to Data” feature description.
- Setting up data calls (or voice calls) for another terminal from a data terminal, which is described in the “Third-Party Call Setup” feature description.

Data Services Overview

System 25's data features provide switched data transmission at up to 19,200 bps (RS-232 interface), and a 212A modem compatible conversion resource capable of handling data at 300 and 1200 bps.

The system provides switched connections between *data endpoints.* These endpoints include data terminals, personal computers, multiport computers, and modems. Data endpoints are either digital data endpoints or analog data endpoints.

Analog endpoints are connected to System 25 voice terminal or trunk port circuits through a modem in the traditional manner. Digital endpoints are connected to System 25 data port circuits on the TN726 Data Line CP. An Asynchronous Data Unit (ADU) is required in place of the modem used with analog endpoints. Section 4 of this manual shows the connections supported and required connecting equipment.

Data calls can be set up between data endpoints. Analog to analog and digital to digital connections are straightforward; calls between analog and digital endpoints are possible only if the system is equipped with a conversion resource (TN758 Pooled Modem Circuit Pack or external modem pool). System 25 data calls from analog endpoints (including those to digital endpoints) are set up in the traditional manner. The calling party should follow the procedures supplied with his/her modem. However, a Modem Request Code must be dialed when calling a digital endpoint.

Call setup from digital endpoints is facilitated by several data features: *Command Mode, Expert Mode, Data Terminal Dialing, Modem Pooling, Third-Party Call Setup,* and *Transfer To Data.*

In the discussion that follows, it is important to understand the difference between analog voice terminology and data terminology. Refer to the "Glossary" (Section 9).

The following provides a definition of a data call in terms of its contextual components. The components are (1) data endpoints, (2) data endpoint states, (3) data call processing modes, (4) connecting configurations, and (5) controlling features.

Data Endpoints

Data endpoints are composed of data terminal equipment, an ADU or modem, and a connection to the switch via an analog or data port. A digital data endpoint is addressed by its Data Dial Code (DDC). Analog data endpoints are addressed like other voice terminals, by their PDCs. For the remainder of this description, data endpoints will refer to digital data endpoints unless stated otherwise.

Several different categories of data endpoints are supported. The categories have been divided into two general groups, those having a DTE type interface, which encompasses almost all of the data terminal devices, and a group of DCE interface devices (primarily modems). The groups have then been divided into categories based upon their functional attributes. However, it must be noted that within each category, control interfaces may vary.

The following describes the categories and attributes of each:

1. DTE Devices

This group of data endpoints have one thing in common: their interface configuration (although RS-232 control signal utilization varies significantly from terminal to terminal). Some data equipment do not use any RS-232 control signals; these require only BA (Transmitted Data Ready-Tx), BB (Received Data Ready-Rx) and AB (Signal Ground) to function, while others require more RS-232 control signals to operate. An ADU (Figure 2-14) can send Data Terminal Ready (DTR) from the data terminal to the Data Line circuit and the Data Line circuit can send a control signal to the data terminal. The signals Data Set Ready (DSR), Clear To Send (CTS), and Received Line Signal Detector (DCD) are all connected to the control signal from the DLC in the ADU and available if required by the data terminal. Refer to Section 5 (Port Specifications) for additional information.

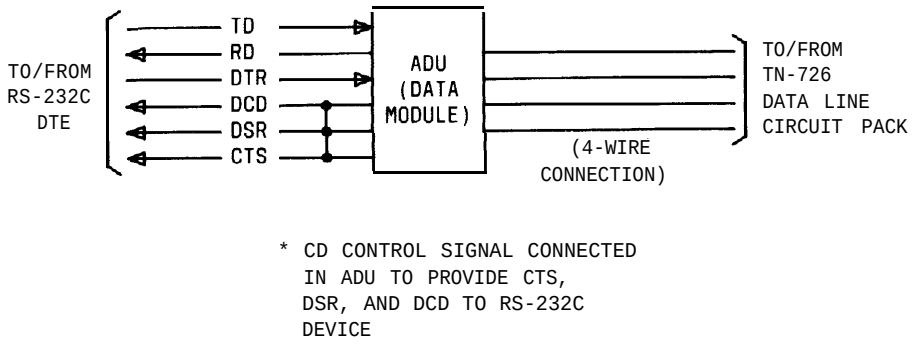


Figure 2-14. Asynchronous Data Unit Interface Signals

The following categories are part of the DTE data endpoint group:

a. **Data Terminal Without ASCII Keyboard**

This category includes such devices as Fax machines, EBCDIC or Baudot terminals, and receive only devices such as printers.

Once connected to an ADU and turned on, these data endpoints appear on-line, available, and ready to enter the Setup mode on auto-answered calls (modes are described below). These endpoints will display or print information received after a valid connection has been established without additional RS-232 control from that endpoint. Note that since these endpoints cannot establish calls for themselves, they must either be called by other endpoints *or* have calls established for them via the Transfer to Data or Third-Party Call Setup feature.

b. **ASCII Data Terminal With Keyboard**

This category can be subdivided into two classes: (1) basic terminals, and (2) intelligent, programmable data equipment such as personal computers. Basic terminals appear to the data port to be on-line and available whenever they are turned on, thus ready to enter either the Data Terminal Dialing mode, or to enter the Setup mode on calls originating from a voice terminal or on auto-answered calls.

In the case of most personal computers, a communications program must be executed in order for it to communicate with its own RS-232 port or built-in modem. Once the communications software is running, further operation will be similar to that of the basic terminal.

c. **Host Computer Endpoint**

A host computer endpoint is very similar to a data endpoint with keyboard except that the host has many ports and the interface is usually capable of supporting multiple speeds and more of the RS-232 control signals. Front-end communication software running in the host is typically supplied by the computer vendor and is not designed to support the Data Terminal Dialing feature. Such software typically supports call origination through Automatic Calling Units (ACUs), which are not compatible with Data Terminal Dialing. Thus, the primary means of communicating with the host is by calling from data terminals or personal computers. Groups of host ports with matching characteristics may be members of hunt groups (referred to as host port groups).

d. **Analog Data Endpoint**

Data endpoints with modems are referred to as analog data endpoints. Modems connected via tip ring lines use PDCs as extension numbers rather than Data Dial Codes (DDCs). Station-to-station data calls to (or from) this endpoint from (or to) digital endpoints require a modem conversion resource to convert the endpoint's analog data to digital format. Calls from a digital endpoint to an analog data endpoint (i.e., calls to a PDC), will automatically have a conversion resource inserted in the calling path. If the called (analog) endpoint should then invoke Transfer To Data, the conversion resource will be released. Data calls originating from an analog data endpoint must first enter a Modem Request Code before addressing a digital data endpoint. This is required because the system assumes that a call originating from a voice terminal will invoke Transfer To Data. If the originating station is not going to transfer to data, it must indicate this so that a conversion resource will be included in the connection.

2. DCE Devices

This group of data endpoints consists primarily of modems. The modems are connected to a data port from their RS-232 side. The modem must be configured as a DTE interface to connect to a System 25 data port. It is possible to simulate a DTE interface from a modem with a cross-over ("null modem") cable. This group of

endpoints is important for users who provide their own modems, connected to dedicated trunks or private lines, for internal modem pooling.

Data Endpoint States

The data endpoint may take on three states: (1) off-line, (2) on-line (on-hook), and (3) on-line (off-hook). Off-line is when a data terminal is out of service (turned off, disconnected, etc,). The on-line (on-hook) state occurs when the terminal is turned on, is available to answer a call, but is not on a data call. Finally, the on-line (off-hook) state is when the data endpoint is actively on a data call.

Data Call Processing Modes

Data calls differ both in signaling and call setup from voice calls. For this reason, a unique set of data call processing modes have been defined to support data call operation in a manner consistent with the characteristics of data terminals.

a. **Off-Line Mode**

The data endpoint is considered to be in the Off-Line Mode whenever the data endpoint's DTR signal is inactive (e.g., "turned off"). The endpoint is considered unavailable and calls to this endpoint will receive the "RINGING" message or Ringback (indefinitely).

b. **Idle Mode**

The Idle Mode indicates that the data endpoint is in its on-line, on-hook state. While idle, call processing will allow the endpoint to:

— Enter either Data Terminal Dialing mode to originate a data call, or enter the Setup mode after a call is originated from a voice terminal (Transfer To Data) or other data terminal (Third-Party Call Setup).

— Autoanswer a data call and go into Setup mode.

The data endpoint remains in the Idle mode while the user is establishing a data call from a voice terminal until Transfer To Data is activated.

c. **Command Mode**

Command Mode enables the Data Terminal Dialing feature, allows the user to view and change associated data port options, and provides access to the Third-party Call Setup feature. Command Mode may be entered by going on-line and pressing Break or Break-Return.

d. **Expert Mode**

Expert Mode is an enhancement to the Command Mode feature that provides an alternative method of performing the full range of Command Mode functions. By eliminating the display of menus and allowing multiple commands to be entered on a single line, Expert Mode lends itself to computer-driven instructions. Individual users who are very familiar with Command Mode operations may also find it useful.

e. **Data Terminal Dialing Mode**

Data Terminal Dialing is a data feature accessed via Command Mode. It provides a procedure to establish data calls without the use of a voice terminal. Data Terminal Dialing supports both on-premises and off-premises data calls (with the support of the System 25 Modem Pooling feature). Dialed digits are entered from the data terminal keyboard or host computer (using a program compatible with Data Terminal Dialing protocol). Call progress text messages are sent to the terminal in place of call progress tones. Upon completion of digit entry, Data Call Setup mode is entered.

f. **Data Call Setup Mode**

Data Call Setup Mode is a transitional state entered after Transfer To Data, Data Terminal Dialing, or during auto-answer; it exists during the handshake between data ports.

If the endpoints are compatible and handshaking is successful, a data connection is established. If handshake failure occurs, the user is notified and the data endpoint returns to the Idle mode. Successful handshake must occur within 15 seconds of answer at the called data endpoint. This implies that the voice terminal user must invoke Transfer To Data within 15 seconds after far-end answer. Similarly, if an originating voice user calls a voice terminal and both users transfer to data, both ends must transfer within the 15 second time limit.

If the data endpoint is optioned for Command Mode permission, the data endpoint will receive call progress text messages while in the Data Call Setup mode.

g. **Data Mode**

Data Mode is first entered after successful completion of Data Call Setup. Transparent communication between connected endpoints is provided in Data Mode.

Connecting Configurations

Refer to “Connectivity” in Section 4 for data equipment connections.

Controlling Features

It is possible to originate data calls from either a voice terminal with a Transfer to Data button or from data endpoints that support Command Mode (i.e., ASCII data terminals with keyboards and host computers). Several controlling features are provided to allow data endpoints and voice terminals to set up data calls. The following briefly describes the Data Service features used in controlling data calls:

a. **Command Mode/Expert Mode**

Command Mode and Expert Mode provide an interface to the Data Terminal Dialing feature, the Third-Party Call Setup feature, and permits users to display and change data port options.

b. **Data Terminal Dialing**

Data Terminal Dialing provides call setup from terminals and host computers.

c. **Transfer To Data**

Transfer To Data is the preferred method of data call origination from multiline voice terminals equipped with Transfer to Data (DATA) buttons and associated digital data endpoints. The DATA button is associated by DDC with a near end data endpoint. A unique DATA button must be provided for each DDC that the voice terminal is capable of controlling. Associated with each DATA button is an LED that reflects the status of data endpoints as follows:

- Dark—Data endpoint is idle
- Winking—Data endpoint is reserved (preindicated)
- Flashing— Data endpoint is being alerted of an incoming call
- On Steady—Data endpoint is either in the on-line (off-hook) state or is reserved for another user and busy.

Refer to the following feature descriptions for additional information:

- Command Mode
- Data Call Setup
- Data Terminal Dialing
- Expert Mode
- Modem Pooling
- Third-Party Call Setup.
- Transfer To Data

Data Terminal Dialing

Description

This feature allows users to originate (place) data calls from a data terminal. Voice terminal dialing and call progress tones are replaced by keyboard dialing and call progress messages. The message “DIAL:” prompts the user to enter the called number from the keyboard, and “RINGING” or “DIALING . . . COMPLETED” informs the user that the dialed number is being called.

Table 2-1 provides a list of the call progress messages.

Note: The following dialing procedures assume that “Command Mode” is active. Refer to the feature description on “Expert Mode” for an alternative method of dialing.

Dialed Characters:

In addition to digits and the “#” and “*” characters on the touch-tone pad, the dialed number may contain the following special characters:

- The characters “(” “)” “-” and SPACE may be used to improve legibility. These characters are ignored.
- The “%” or “,” characters may be used to cause a 1.5-second pause in dialing. Multiple pause characters can be used to produce longer pauses.
- The “$” (mark) character indicates that the remaining digits are for end-to-end signaling.
- UNDERSCORE or BACKSPACE characters may be used to correct previously typed characters on the same line.
- The “@” character may be used to delete the entire line and start over with a new DIAL: prompt.

Each line of dialing information may contain up to 27 characters. Note that all of the dialing information, including pauses and ignored characters, must be typed on a single line following the DIAL: prompt and terminated by Return.

Dialing Correction:

The backspace character (BS key or Ctrl-H keys) or underscore (“_”) may be used to cancel the previously entered character. More than one entered character may be deleted by using multiple backspace or underscore characters. The “@” character may be used to delete the entire line of entered characters.

Pause:

To assist the completion of off-premises calls, the pause characters “%” or “,” may be used. A pause character may be used to help ensure the receipt of dial tone before continuing to dial. Each “%” or “,” causes a fixed delay of one and one-half (1.5) seconds.

Pause characters may be used consecutively if a longer pause is required. Note that System 25 cannot detect tones, such as a second dial tone for end-to-end signaling.

End-to-End Signaling:

Data connections to off-premises destinations require that a conversion resource (pooled modem) be inserted into the connection. Occasionally it is necessary to send additional tones to the remote endpoint after the connection is established to signal the remote equipment. A "mark" character ($) must be included on the DIAL: line to indicate to call processing that the remaining digits are to be sent to the far end prior to insertion of the conversion resource into the connection. The "mark" character marks the boundary between the digits dialed to reach a distant endpoint, and the digits used by that distant endpoint after it has answered. Pause characters may and usually should follow a "mark" character. An example using a "mark" character and several pause characters is shown below. Dashes are included for readability.

Examples of dialing are as follows:

- DIAL: 3478
- DIAL: 9-1-(201)-946-8123,,$,5678
- DIAL: 9%946-8123%%$%5678%137%110

Call Disposition:

Call progress messages corresponding to call progress tones provided to voice terminals are listed in Table 2-1. The message supplied (indicating reorder, busy, ringback) depends on the disposition of the call.

1. When ringback is received the displayed message is "RINGING" (internal calls only). For outside calls, the corresponding call progress message is "DIALING . . . .".

2. If the endpoint answers, the displayed message is "ANSWERED" (internal calls only). Then, if the handshake succeeds, a data connection is established. For outside calls, when the system has finished dialing, the message "COMPLETED" is displayed.

3. If the handshake fails because a connection cannot be established between endpoints (e.g., a port optioned at 9600 baud attempts to talk to a conversion resource that can only talk at 300 or 1200 baud), the user receives "INCOMPATIBLE FAR END, " " DISCONNECTED, " and the data endpoint goes on-hook.

If the far end does not answer, the caller must press Break to terminate the call attempt.

If the disposition of the call is such that TRY AGAIN or BUSY (indicating reorder or intercept and busy respectively) is received, the switch sends "DISCONNECTED" to the data terminal and returns the data endpoint to idle mode.

Table 2-1. Call Progress Messages for Data Terminal Dialing

Displayed Message	Application	Meaning
DIAL:	Placing a call	Equivalent to dial tone. Enter any required facility number followed by the dialed number and a RETURN.
RINGING	Placing a call	Equivalent to Ringback Tone. Called number (far-end) is being signaled. Provided on internal calls only.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use, or out of service. Provided on internal calls only.
ANSWERED	Placing or receiving a call	Notifies calling and called users that call has been answered. Provided on internal calls only.
TRY AGAIN	Placing a call	Equivalent to Reorder Tone. System facilities are currently not available or invalid number.
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
PLEASE ANS-	Placing a call from a voice terminal	Originating voice terminal user has transferred call to data terminal using Transfer to Data.
DISCONNECTED*	Call is terminated	Call or call attempt is disconnected.
CONNECTED, SPEED = NNNN	Call is connected	Notifies user that the call connection is established and what the baud rate is. [Provided that “Connection indication” (Data Port Action 77) is enabled.]
INCOMPATIBLE FAR END	Placing a call	Notifies user that the handshake between data end points has failed
DIALING . . COMPLETED	Placing a call	Indicates off-premises call is being dialed and that dialing is completed.
PLEASE WAIT	Placing a call	Call queued.
BAD NUMBER	Placing a call	Bad dialed number
NO MODEM	Placing a call	No modem available for a call that requires one.
SESSION 1	Placing or terminating a call	Specifies the session number (1) of the data call to the calling party

* Bell sounds when message is displayed.

Answering Endpoint:

When the dialed endpoint is alerted, the user receives “INCOMING CALL-”. (The called terminal will auto-answer if it is turned on.) If the handshake succeeds, a data connection is established and the “CONNECTED” message is displayed if so optioned. If the handshake fails, the user receives “INCOMPATIBLE FAR END, DISCONNECTED” and the data endpoint returns to idle mode.

Considerations

Data Terminal Dialing allows users to place data calls from their terminals using the Data Terminal Dialing feature and allows users to review the options administered for their data ports.

Interactions

The following features interact with Data Terminal Dialing

End-To-End Signaling: (See preceding text.)

Modem Pooling: Data calls between analog and digital endpoints require that a conversion resource (TN758) be available. If one is not, the “NO MODEM” followed by “TRY AGAIN” message will be displayed.

Speed Dialing: System Speed Dialing codes can be dialed from data terminals. Personal Speed Dialing is not supported.

Administration Requirements

Data Port: See the table of Permissible Data Port Options in the “Command Mode” feature description.

Hardware Requirements

TN726 Data Line CP to support each digital endpoint.
TN758 Pooled Modem CP to support data calls between digital and analog endpoints.

Dial Access to Message Waiting Indicators

Description

This feature allows users to turn on or turn off the message waiting indicator on any voice terminal in the system by dialing a code.

To turn on a Message LED at some station, the user first goes off-hook, or flashes the switchhook, to get dial tone. The user then dials activation code #90 followed by the extension number of the target station. If the attempt to turn on the LED is allowed, the caller receives confirmation tone, and the connection is dropped. If the dialed station has no Message LED or if the extension number is invalid, the attempt is denied and the caller gets reorder tone.

In conjunction with Dial Access service, each Direct Group Calling (DGC) group in the system may have one station assigned as receiver of message waiting indications. If a caller dials the number of a DGC group, the system routes the message waiting request to the extension of the designated message waiting indication receiver.

The procedure for turning off a Message LED parallels the turn-on procedure. The user gets dial tone, then dials deactivation code #91 and the extension number of the target station. Confirmation tone is returned if the attempt is successful, reorder tone if it is not.

Considerations

Dial Access to Message Waiting Indicators provides users with a way to notify any other terminal that a message is waiting.

This feature does not apply to the feature buttons/LEDs administered for Station-to-Station Message Waiting.

Interactions

Dial Access to Message Waiting Indicators can coexist with the other messaging services in System 25. Careful management is essential so that users know where to retrieve their messages.

The following feature interacts with Dial Access.

Display: When a display set user dials #90 (or #91) followed by an extension number to light (or extinguish) a Message LED at some station, the dialed digits are displayed. A confirmation of Message LED activation or deactivation is not displayed.

Hardware Requirements

Only terminals with built-in message waiting indicators (designated MSG or MESSAGE) or Z34A Message Waiting Indicator adjuncts can be signaled by this feature.

Dial Plan

The dialing plan for System 25 is based on the concept that, whenever possible, calls should be placed to individuals rather than to voice terminals. To implement this concept, individuals are assigned Personal Dial Codes (PDCs) and are allowed to sign in those PDCs at other voice terminals. There are two types of PDCs: assigned and floating. An assigned PDC is associated with each voice terminal. Floating PDCs (FPDCs) may be signed in at any voice terminal. Calls to FPDCs will ring at the signed-in terminal and may (optionally) ring at the attendant position when not signed in anywhere.

Data extensions on System 25 are assigned Data Dial Codes (DDCs).

Dial Code Assignments

System 25 dial codes are as follows:

Assignable System 25 dial codes may have 1, 2, 3, or 4 digits. These include voice terminal PDCs, data terminal DDCs, Direct Group Calling (DGC) Groups, Paging Access, Attendant (Selector Console) Park, Night Service, Modem Request, Automatic Route Selection Access, Facility Access (trunk group), and Dictation System Access codes.

System 25 fixed dial codes are:

- 0—Attendant access
- * 1, *2, *3—Reserved
- *4—Activate Make-Busy for DGC group member
- *5—Park
- *6—Deactivate Make-Busy for DGC group member
- *7 + 0—Group Pickup Answer
- *7 + PDC—Directed Pickup Answer
- *8 + PDC—Park Retrieval
- *9—Camped-On/Call Waiting Call Retrieval
- *0—Account Code Entry
- ** PDCPDC—Sign in PDC (Following and Forwarding)
- ** PDC0—Sign out PDC (Following)
- **0—Sign out all PDCs (Following)

- *—PAUSE character (in programmed numbers)
- #100-#189—System Speed Dialing Codes
- #190-#199—Virtual Facility Codes
- #20-#39—Personal Speed Dialing Codes
- #4—Activate Program mode
- #5—Insert dialed digits here (in Virtual Facility numbers)
- #8—Start end-to-end signaling (in programmed numbers)
- #60-Activate Callback Queuing at single-line voice terminal
- #61—Cancel Callback Queuing request at single-line voice terminal
- #70—Activate Forwarding
- #90—Activate Dial Access to Message Waiting Indications
- #91 —Deactivate Dial Access to Message Waiting Indications
- #92—Activate Leave Word Calling
- #—End of dialing
- ##—Sends a "#" (in programmed numbers)
- #* —Sends a "*" (in programmed numbers)
- ## PDC—Call Accountability.

The maximum number of dial codes available for a System 25 is 600. Each assigned code is stored individually in memory.

The dial codes assigned in the system must be completely unambiguous. For example, a dialing plan that contains the number "20" cannot contain the numbers "2," "200-209," or "2000-2099."

PDC to Voice Terminal Association

During installation, each voice terminal is assigned one PDC that serves as its extension number. These are referred to as "assigned" PDCs, and the associated terminals are called home stations. Additional "floating" PDCs (FPDCs), may be assigned in a system. At the customer's option, floating PDCs may have the attendant position assigned as their home station (i.e., calls to FPDCs will be directed to the attendant when they are not signed in anywhere). A maximum of 200 assigned PDCs and 300 FPDCs may be allocated in a system.

Data Dial Codes (DDCs)

At the time of installation each digital data endpoint will be assigned a Data Dial Code (DDC) that serves as its extension number. A maximum of 104 DDCs may be allocated in a system.

Direct Inward Dial (DID) Number Assignments

Each DID number is associated with a unique PDC (floating or assigned), a DGC access code, a Remote Access point, a DDC, or a pooled facility access code. The code associated with a DID number is the last 2, 3, or 4 digits of the DID number. For example, the code associated with the DID number “NXX-2157” will be 57, 157, or 2157.

All dial codes in the system that are associated with DID numbers should have the same number of digits. However, there is no requirement that all PDCs, DDCs, DGC access codes, or facility access codes be associated with DID numbers.

Voice Terminal Directed Features

Directed Night Service, DGC calls, Personal Line Calls, Manual Signaling, Station Message Waiting, Automatic Intercom, and Outward/Toll Restriction are associated with specific terminals (stations), not with PDCs. This means that these features do not move with a PDC when it is signed in at another voice terminal.

Dictation System Access

Description

This feature permits voice terminal users to access and control customer-owned dictation equipment. System 25 can provide an interface to dictation systems that require either an industry-standard station line port or an Auxiliary Trunk port with contact closure (equivalent to a push-to-talk switch).

The dictation system is accessed by dialing the designated access code or by pressing a DSS button on which this access code is stored.

Considerations

This feature allows users to access and control a shared dictation system. A dial dictation system is essentially a sophisticated tape recorder that can respond to touch-tone signals for physical control. For instance, pressing the digit six might cause the dictation system to rewind its tape; pressing the digit three might cause the dictation system to play back its tape.

Most modern dial dictation systems interface to System 25 through an industry-standard station line port. However, some dictation systems require contact closure for recording control and must interface to System 25 through a port on an Auxiliary Trunk CP (TN763) by means of an Auxiliary Trunk Interface and a Paging/Dial Dictation Interface.

If a dictation system may be optioned for either of these interfaces, the preferred interface is the station line port.

Interactions

The following feature interacts with Dictation System Access.

Direct Inward Dialing (DID): A DID number may match the dictation system access code. This allows an outside caller to access the dictation equipment.

Administration Requirements

System:

- Dial dictation equipment requires a suitable port to interface to System 25. A port on a ZTN78 Tip Ring Line or TN742 Analog Line CP is the preferred interface if the dictation equipment can be optioned for a station port. A port on a TN763 Auxiliary Trunk CP and its associated equipment must be used if the dictation system requires a separate contact closure for proper operation.

Voice Terminal Port:

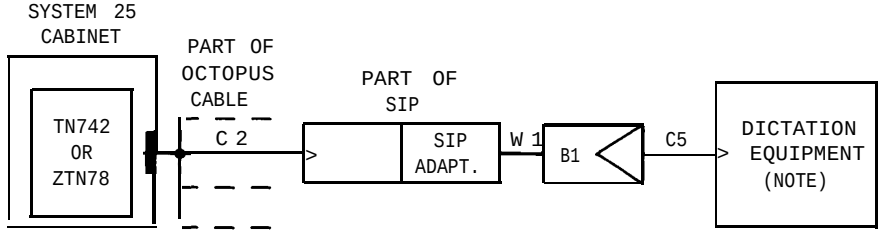
- Assign DSS access buttons, as desired.

Hardware Requirements

Customer-provided dictation equipment suitable for connection to a telephone system.

Port on a ZTN78, TN742, or TN763 CP, as required. If the equipment requires a contact closure, the TN763 and supporting equipment must be used. Detailed connection information is provided in Figure 2-15.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters are provided under the heading "Connectivity" in Section 4 of this manual.



LEGEND:

TN742 – ANALOG LINE CP
ZTN78 – TIP RING LINE CP
B1 – TYPICAL- 103A CONNECTING BLOCK*
C2 – OCTOPUS CABLE (WP90780) - PEC 2720-05P
C5 – MODULAR CORD (D4BU-87)
W1 – 4-PAIR INSIDE WIRING CABLE*

* – FURNISHED BY INSTALLER

NOTE: IF CUSTOMER DICTATION EQUIPMENT REQUIRES A CONTACT CLOSURE, A TN763 AUXILIARY TRUNK CP MUST BE USED. REFER TO THE “PAGING SYSTEM ACCESS” FEATURE DESCRIPTION FOR TYPICAL CONNECTION INFORMATION.

Figure 2-15. Dictation System Connections (FCC Registered)

Digital Tape Unit (DTU)

The Digital Tape Unit (Figure 2-1 6) is a RS-232 device used to record administration translations. The DTU does not encode the translations data as it records, nor does it require decoding circuitry when playing back (restoring) recorded data. Data is recorded and transmitted at 1200 bps.

The DTU requires 115V commercial power from a 3-wire grounded outlet. It should be located on a desk or table top. The recorder is approximately 5 inches wide, 2 inches high, and 10 inches long.

As shown in Figure 2-17, the DTU must be directly connected to port #3 on the Call Processor (ZTN82 or ZTN128) CP. Remote and switched connections are not supported. Maximum cabling distances are provided in Section 5, “Technical Specifications. ”

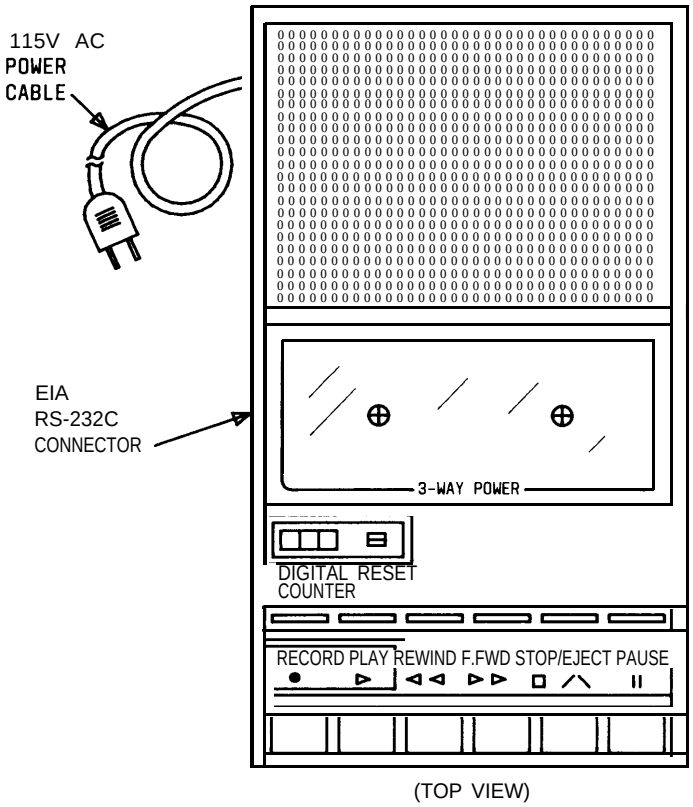
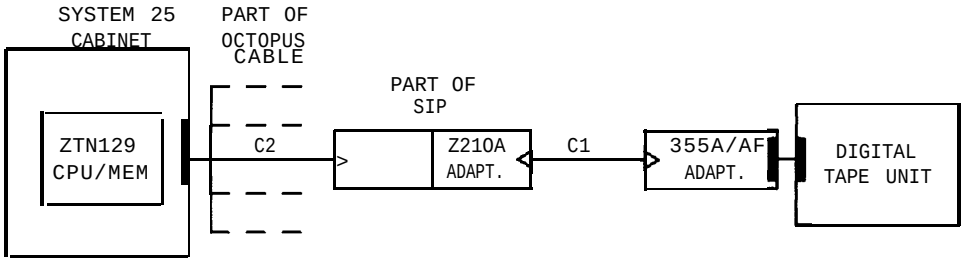


Figure 2-16. Digital Tape Unit



LEGEND:

C1 – MODULAR CORD (D8W-87) - PEC 2725-07G
C2 – OCTOPUS CABLE (WP90780) - PEC 2720-05P
355A ADAPTER – RS-232 PLUG TO MODULAR JACK - PEC 2750-A24
355AF ADAPTER – RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-17. Digital Tape Unit—On-Premises Direct Connections (Sharing Same AC Outlet)

Direct Group Calling (DGC)

Description

Direct Group Calling (DGC) allows incoming calls to be directed to a specific group of terminals. Up to 32 DGC groups, each including up to 20 members, may be set up. A terminal can be in only one DGC group. Incoming calls on particular trunks can be directed to a DGC group. These trunks can also be used for outgoing calls.

DGC groups can be administered as Coverage Groups receivers. Refer to the “Coverage, Group” feature description for details.

Calls to a DGC group hunt in a circular manner, starting at the terminal following the last one to receive ringing (whether answered or not), and will ring at the next idle terminal in the group. On multiline voice terminals the calls arrive on a SYSTEM ACCESS button (LOOP button on a Switched Loop Attendant Console).

If all group members are busy (off-hook), an outside call is queued and the caller receives ringback tone. If the system includes a delay announcement, it is played after a specified number of rings. The caller is subsequently put on hold (in queue) and will receive Music-On-Hold if available. If the system is not equipped with a delay announcement, the call will begin to ring at all line appearances after the specified interval.

An inside caller dials a DGC access code to reach a DGC group. If all members of the group are busy, the call will go into a queue if Callback Queuing is activated either automatically or manually, otherwise, the call will not queue and the caller will receive Busy Tone.

Once the call begins to ring at a group member’s station, it will **not** receive announcement service or ring at a line appearance. For this reason, it is important that DGC members log out (as described below) when they will be away from their desks.

The attendant can camp-on multiple outside (trunk) calls when all members of the group are busy. Group members do not receive camp-on indication. The camped-on calls will be queued, and are eligible for the DGC delay announcement. If no delay announcement is available, the calls will return to the attendant console after a specified number of rings.

DGC group members may withdraw from the group (log out) by going off-hook and dialing *4. To reenter the group (log in), the member goes off-hook and dials *6.

An off-hook multiline terminal or attendant console (even if busy on only one SYSTEM ACCESS or LOOP button) appears busy to DGC calls. However, terminals other than the SLAC may receive other (non-DGC) calls while active on a DGC call.

Direct Group Calling groups may be used for data applications to access host ports and the STARLAN Interface CP. The System Administrator may disable queuing for data DGC groups, if desired. Delay announcements and music-on-hold are not provided for data groups.

Considerations

DGC groups are particularly useful when the answering group receives a high volume of calls. Call completion time is minimized and attendant assistance is not required.

Any number of outside trunks may be administered to feed into a DGC group. A trunk may feed only one DGC group.

Interactions

The following features interact with Direct Group Calling.

Attendant Console, Switched Loop: When an incoming trunk call rings simultaneously at a DGC queue and a Switched Loop Attendant Console queue, it may be answered by either, depending on who answers first.

Attendant Direct Extension Selection: When all stations in a DGC group are busy, the status LED on the Selector Console lights steadily.

Bridging of System Access Buttons: DGC calls arriving on System Access buttons at a principal station can receive bridging treatment at a bridging station.

Callback Queuing: Inside calls to busy DGC groups can be queued. Queuing is not allowed if all members of the DGC group are logged out. A multiline DGC member with a queued call and a single-line member with an off-hook queued call are considered busy.

Coverage: When a call rings at DGC station that has Coverage, the call will receive that station's coverage. Calls directed to a busy DGC group do not receive coverage. Instead, after a predefine number of rings, a trunk call will be transferred to a delay announcement (if provided), or ringing will be transferred to all button appearances of the line and the SLAC queue (if trunk has ringing enabled).

Display: A logged-in Direct Group Calling (DGC) group member can view the number of calls waiting to be serviced by the group. The display is continuously updated for all members. A digit, 0 through 9 or "!" for 10 or more, appears in position 16. DGC queue values are not displayed at a SLAC assigned to a DGC group; the attendant's display always contains the number of calls waiting in the attendant queue.

Direct Group Calling Delay Announcement: Provides a recorded announcement to an outside (trunk) caller who has been placed in queue for a DGC group.

Direct Inward Dialing: An incoming DID call may match a DGC group access code.

Direct Station Selection (DSS): A DSS button can be assigned to a DGC group. The associated LED lights steadily when all stations in the group are busy.

Modem Pooling: Modem Pooling supports calls to data endpoints that are part of a DGC group. While an incoming data call is in a DGC group queue, the caller receives ringing. The conversion resource is inserted if the call is completed to a digital endpoint.

Personal Lines: An outside line directed to a DGC group can be assigned button appearances in addition to the DGC group assignment. When an incoming call is ringing at a DGC group, the status LED on the voice terminal button appearance lights steadily, indicating that the line is busy. If the call goes unanswered for a pre-determined number of rings (and no delay announcement is provided), ringing will be transferred to all button appearances of the line and the status LED will flash.

Pickup: A DGC group member can also be a member of a Pickup group.

Remote Access: Remote Access callers cannot log into or out of a DGC group.

Station Message Detail Recording (SMDR): For an incoming call to a DGC group that was connected to an announcement but was never answered, “0” will be reported in the “STN” field of the call record. If the call was answered by a station after receiving the announcement, that station will be listed in the ‘‘STN” field.

Tie Trunks: Calls to a busy DGC group via tie trunks will be queued and will receive a delay announcement, if available.

Transfer: Internal stations can transfer outside (trunk) calls to a busy DGC group. The transferred call will be treated as any other trunk call to a busy DGC group. The transferring party will hear Busy tone, but the transfer will complete. The call will queue and the calling party will receive delay announcement, if available.

Administration Requirements

Trunk Ports:

- Assign trunks to DGC Group.
- Assign trunks to ring in SLAC queue.

System:

- Assign number of rings before DGC calls are transferred to announcement or begin ringing at button appearances or SLAC queue.

Direct Group Calling:

- Assign DGC access code, add/delete DGC members, enable/disable queuing for data DGC groups.

Direct Group Calling Delay Announcement

Description

This feature provides a recorded announcement to an *outside* (trunk) caller who has been placed in queue for a DGC Group.

When all members in the group are busy (off-hook), the call will be queued for DGC service and the calling party will receive ringback tone. Note that no incoming call indication (ringing) is provided to the DGC group members at this point. After a specified number of rings (administrable) a recorded announcement will be played to the calling party without disturbing his or her position in queue. The caller is subsequently placed on hold and will receive music if available.

Once a call begins to ring at a DGC station, the call is no longer eligible for delay announcement service. The call will then ring until answered, covered, picked up, or abandoned.

Considerations

DGC Delay Announcements provide the calling party with a message that acknowledges their call and assures them that their call will be handled in an orderly way.

Interactions

The following feature interacts with Direct Group Calling Delay Announcement.

Tie Trunks: Calls to busy DGC groups via tie trunks will be queued and will receive the delay announcement, if available.

Administration Requirements

The DGC announcement device requires a port assignment on a ZTN78 Tip Ring Line or TN742 Analog Line CP. Only one DGC Delay Announcement may be assigned in the system. Callers to all DGC groups receive the same message.

Hardware Requirements

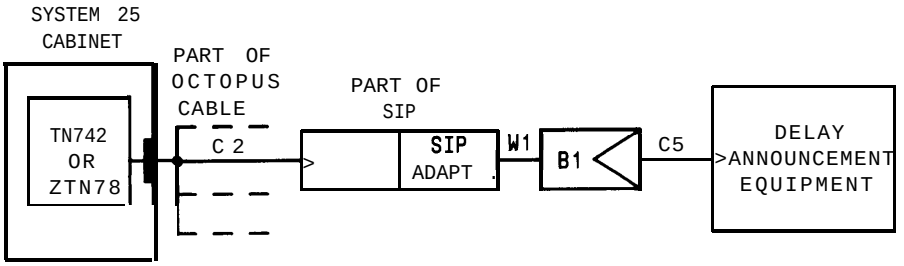
The announcement device *must* automatically hang up at the end of each call so that the incoming call can be returned to the DGC queue.

The equipment requires a port on a ZTN78 Tip Ring Line (or TN742 Analog Line) CP. The system supports one DGC delay announcement.

For Music-On-Hold hardware information, refer to the “Music-On-Hold” feature description.

Detailed connection information is provided in Figure 2-18.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” in Section 4.



LEGEND:

TN742 - ANALOG LINE CP
ZTN78 - TIP RING CP
B1 - TYPICAL-103A CONNECTING BLOCK*
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C5 - MODULAR CORD (D4BU-87)
W1 - 4-PAIR INSIDE WIRING CABLE*

* – FURNISHED BY INSTALLER

Figure 2-18. Delay Announcement Equipment Connections (FCC Registered)

Direct Inward Dialing (DID)

Description

Direct Inward Dialing (DID) allows incoming dial pulse type calls to reach specific individuals or facilities in the system without attendant assistance.

System 25 customers reserve blocks of DID numbers from the CO. The DID numbers may correspond to a PDC, FPDC, DGC access code, DDC, or any facility with an access code such as a pooled facility or a paging zone.

The system is capable of receiving either 1, 2, 3, or 4 digits over its DID trunks. The number of digits received on a specific DID trunk will be constant for that trunk; however, different DID trunks may receive different numbers of digits. The system is capable of receiving up to four digits and then ignoring leading digits as specified to match against system dial codes. For example, the dial code matching DID number NXX-2157 can be 57, 157, or 2157. If the System 25 is administered to match on more digits than are received from the Central Office (CO), the additional leading digits are taken from the 4-digit trunk number. For example: if a call comes in on DID trunk number 1234, the CO sends two digits (77) over this trunk to identify the recipient, and System 25 is administered to match on three digits, then the call will be routed to dial code 277.

incoming DID numbers that don't match any valid dial code may optionally be directed to the Attendant Console or to Reorder Tone.

If the DID number received is a valid dial code, the caller is provided either Ringback Tone, Busy Tone, or the tone from a pooled facility (e.g., Dial Tone) as soon as addressing is completed. Busy Tone is provided if and only if the call cannot be completed to the intended voice terminal and cannot be provided coverage.

DID calls appear at System Access buttons on multiline voice terminals (they do not have other button appearances). These calls can be transferred to a covering station, answered via Pickup, directed to a DGC Group, or given Station Hunting, Following, or Forwarding treatment. A DID call *is not* automatically covered on the Attendant Console.

DID trunks may utilize DID Immediate Start or Wink Start protocols, but must be dial pulse (touch-tone DID trunks are not supported). Refer to Section 9, Glossary, for a brief description of each of these trunk types.

Considerations

Direct Inward Dialing frees the attendant from handling certain incoming calls.

Interactions

The following features interact with Direct inward Dialing.

Attendant Camp-On: DID calls are not provided Attendant Camp-On treatment. They will not appear on the Direct Trunk Attendant Console Return-On-Busy or Return-On-Don't-Answer buttons or on the Switched Loop Attendant Console Loop buttons *unless* they are first answered at the attendant position and are subsequently extended by the attendant.

Attendant Direct Extension Selection: Selector Console LEDs respond to DID calls just as they do for other outside calls. When a user answers a DID call, the associated LED on the Selector Console will light steadily. When a DID call arrives at the attendant position for coverage, the LED associated with the coverage sender will flash and will then go dark when the call is answered. However, if the call is placed directly to the attendant position or is forwarded to the position and thereby arrives on a System Access button or a Loop button (e.g., if a DID PDC is signed-in at the attendant position), then no LED indications on the Selector Console will be provided. If a DID call is directed to the answering position and is subsequently extended to a station, then the LED on the Selector Console associated with the station will flash if the call returns to the answering position. The LED will light steadily if the call is answered by the station.

Call Waiting: Incoming DID trunk calls do not wait at busy stations; they receive busy tone.

Conference: For conference purposes, DID calls count as one of two allowable outside parties.

Coverage: DID calls receive standard coverage treatment.

Dictation System Access: A DID number may be associated with the dictation system access code. This allows an outside caller to access the dictation equipment.

Direct Group Calling: A DID call will be directed to a DGC group if the DID number matches the DGC group access code.

Night Service: DID calls *do not* receive Night Service treatment. A DID call will ring at the appropriate station whether Night Service is activated or not.

Off-Premises Stations (OPS): DID calls can be directed to OPS.

Paging System Access: A DID call may access a paging zone. This allows the user to dial in and utilize the Paging feature. Dial restricting the paging code will block this interaction.

Personal Dial Codes: DID calls will be redirected to PDCs signed in at other terminals in the system. DID calls to an unassigned PDC or a FPDC that is not signed in will be either redirected to the attendant or receive Reorder Tone.

Pooled Facility Access: Access to pooled facilities via DID is permitted. This includes access to WATS, FX, tie trunks, private lines, dictation equipment, and paging systems. This access is provided by selecting facility access codes so that they will match DID numbers.

Remote Access: A valid DID number can be assigned for Remote Access calls into the system.

Station Message Detail Recording (SMDR): Only one SMDR record is produced if an outgoing call is originated by a DID trunk. The STN field will contain the DID trunk's 4-digit number, the FAC field will contain the facility access code of the trunk group used to complete the call, and the CALLED NUMBER field will contain the called number.

Administration Requirements

System:

- Send calls for unassigned DID numbers to the Attendant Console (yes or no; default = yes),
- Set number of DID digits matched against dial codes (2-4, none; default = 3).

Trunk Port:

- DID trunk type (Immediate Dial, Wink Start)
- Number of digits to be received from CO on this trunk; default = 3.

Hardware Requirements

Each DID trunk requires a port on a TN753 DID Trunk CP.

Directory

Description

This feature allows the user of a voice terminal where the Display feature is administered to search the system's integrated directory data base for the extension numbers associated with specific names. Information resulting from the use of Directory is displayed on the voice terminal's screen.

The user enters Directory Mode from Normal Mode by pressing the DIRECTORY button. The system presents the following display to prompt the user to enter a name using the dial pad buttons and then dial # to mark the end of the search entry.

Screen 1 | DIR: ENTER NAME#

The dial pad buttons are labeled with all the necessary entry characters except as follows:

- Q is entered by pressing button 7 (PRS).
- Z is entered by pressing button 9 (WXY).
- Space, comma, or dot is entered by pressing the * button.

The directory prompt, DIR: ENTER NAME#, remains displayed on the screen until the user finishes entering the characters of the name and presses #. (It is often unnecessary to enter a full name, but whatever is entered must be terminated by #.) The system then searches the directory data base for a match between the entered characters and the stored names. If none is found, the prompt is removed and NO MATCH FOUND is displayed. Otherwise, directory information is presented as shown in the following example:

Screen 1 | D645 Wiggins,G

The D in position 1 indicates that the Directory Mode is active.

If the name is not the correct one, the NEXT button allows the user to request that the next matched name be displayed. This operation can be repeated. As an alternative, the user can narrow the search by entering additional letters followed by #. The additional letters are added to the end of the previously-entered search string. If a mistake is made, the user can press DIRECTORY twice (to exit and reenter the mode) and try to enter the desired name again.

If the user reaches the end of a matched list, NO MATCH FOUND is displayed. The user can return to the first matched name by pressing NEXT again.

When the displayed name is the correct one, the user can call the number by pressing the CALL button. If the terminal is on-hook, the speakerphone will turn on automatically.

The user of a non-attendant display set can exit from Directory Mode directly to Program Mode by moving the program switch on the left side of the terminal to position P. However, to reenter Directory, the user must first go from Program to Normal and then press DIRECTORY.

The user can return from Directory Mode to either Normal Mode or Local Mode by any of the following actions:

- Press DIRECTORY again.
- Allow timeout to occur after 15 seconds with no operation of other buttons (such as NEXT).
- Change switchhook state; if the user goes on-hook, the terminal returns to Normal or Local Mode; if the user goes off-hook, the terminal returns to Normal Mode.
- Select a call appearance button; the terminal returns to Normal Mode.

Considerations

The Directory feature is most effective if the system administrator enters names in the data base in a last name/comma/first initial format. Characters other than letters and numbers (and commas) are discouraged. However, the system does not enforce these rules. A maximum of eleven characters can be entered for a name in the data base, but only nine can be displayed.

Activating the Directory Mode has the following impact on terminal operation:

- Hands-Free Answer on Intercom (HFAI) is disabled.
- If the user is on-hook but has a call on hold, there is no effect on the call.
- If the user is off-hook and in the midst of dialing, the system disconnects the call.
- If the user is off-hook and has completed dialing, there is no effect on the call.
- Automatic Incoming Call Identification (see the “Display” feature description) is suppressed.
- Calls can be originated only by using the CALL button.
- Incoming calls ring and flash, but answering a call will change the terminal from Directory Mode to Normal Mode.

Administration Requirements

Administration of the Display feature enables Directory.

Hardware Requirements

The Directory feature can be used only at display-equipped multiline voice terminals .

Direct Station Selection (DSS)

Description

Direct Station Selection (DSS) allows one-button access to another voice terminal, a paging zone, or a DGC Group. DSS requires a button assignment on a multiline voice terminal.

There are two types of DSS buttons. Numbers stored on DSS buttons (maximum of four digits) are programmed at the SAT; numbers stored on *Flexible DSS* buttons (maximum of four digits) are programmed at the voice terminal. The procedure for programming FLEX DSS buttons is provided in the "Program" feature description.

To use DSS, the user presses DSS or FLEX DSS and goes off-hook. The caller hears Ringback Tone. DSS calls to a multiline voice terminal are received on a System Access button. The DSS status LED is lighted steadily at the calling station.

The DSS status LED is lighted whenever the pointed-to station is off-hook. The user may press DSS and remain on-hook to receive the busy-to-idle reminder. The user's voice terminal will ring once when the other party hangs up; lifting the handset will automatically place the call.

When Prime Line Preference is assigned to a DSS button, the button must be pressed to invoke the busy-to-idle reminder, even though the l-Use LED is lighted.

Access to Paging Zones and DGC Groups:

DSS (not FLEX DSS) access is provided to an individual Paging Zone or to all paging zones or to a DGC group. If the paging zone(s) is administered to be dial restricted, users assigned DSS buttons with paging access codes can still access the paging equipment. The status and busy-to-idle reminder indication described above also apply to DGC groups with the understanding that a DGC group is busy if all members in that group are busy.

Considerations

Direct Station Selection differs from Automatic Intercom in that it provides one-button access from one voice terminal to another (one-way only), while Automatic Intercom provides similar access for each voice terminal (two-way) and must be assigned between two multiline voice terminals. A DSS button may point to a single-line station; an Automatic Intercom button may not. DSS calls receive coverage, Automatic Intercom calls do not.

Interactions

The following features interact with Direct Station Selection.

Bridging of System Access Buttons: Calls from DSS or FLEX DSS buttons on the principal station are not accessible from Bridged Access buttons on the bridging station.

Coverage: DSS calls placed to an individual with Coverage will receive standard coverage treatment.

Display: Operation of a programmed FLEX DSS button generates a display of the information stored on the button; if the button is not programmed, NO INFORMATION is displayed.

Direct Group Calling: A DSS button can be assigned to a DGC group. The associated LED lights steadily when all stations in the group are busy.

Following/Forwarding: DSS calls do receive Following or Forwarding treatment.

Last Number Dialed: Numbers called by pressing FLEX DSS or DSS buttons are not saved by Last Number Dialed and cannot be redialed with that feature.

Line Selection (Prime Line Preference): When Prime Line Preference is assigned to a DSS button, the button must be pressed to invoke the busy-to-idle reminder, even though its red I-Use LED is lighted.

Personal Dial Code (PDC): An attempt to program a FPDC on a FLEX DSS button (rather than a PDC) results in Reorder Tone.

Pooled Facilities: A pooled facility access code may be stored on a FLEX DSS button (but *not* on a DSS button). If so, the button will function very much like a Direct Facility Access button, with the capability of receiving a busy-to-idle reminder for the pooled facility. However, this button will not allow access to a dial-restricted facility.

Administration Requirements

Voice Terminal Port:

- Assign DSS and/or FLEX DSS buttons.

Display

Description

This feature provides visual alphanumeric call information at multiline voice terminals equipped with display modules. The Display feature also provides support for the Directory, Inspection, Local Display, and Program features, which are all covered in separate feature descriptions. Display capability is based on the system's integrated directory, which allows names to be associated with Personal Dial Codes (PDCs), Data Dial Codes (DDCs), Direct Group Calling (DGC) groups, and trunks.

Call information is presented on the 16-character screen located in the upper right area of the following sets. (Refer to Section 4, "Hardware Description, " for complete information and pictures.)

- Model 7305H04C 34-Button Multiline Voice Terminal.
- Model 7317H01A 34-Button Multiline Voice Terminal (BIS 34D).

Both of these terminals can be assigned as general user positions or as Switched Loop Attendant Consoles (SLACs). Display operation is basically the same in both applications; differences will be pointed out in the following descriptions.

The following types of data are presented for calls handled at display terminals:

- The extension number and name of an inside party called from the console
- The extension number and name of an inside party calling the console
- Trunk identification on incoming trunk calls
- Digits dialed on outgoing trunk calls
- Called and calling party information on coverage/redirected calls
- Called and calling party information on returning and third-party calls.
- Call type and reason for return or redirection
- Number of calls waiting in attendant, callback, and DGC queues
- Special information resulting from feature button operation.

Display Screen

The Display feature can generate up to two screens of call information, each of which contains 16 character positions. Screen 1 is automatically activated on incoming and outgoing calls and some feature button operations; no action is required of the station user to access this display. Screen 2 is available for secondary and overflow information about calls.

Display Operation Modes

Most normal call handling activity, such as placing and answering calls and using features, takes place in "Normal Mode." Call displays in this mode require no manual action by the user except for operation of the SCROLL button to display Screen 2 in certain types of calls. Some terminals (SLACs and logged-in DGC stations) remain in Normal Mode unless their users deliberately enter another mode, while others revert to a clock/calendar display (Local Mode) when there is no call handling activity.

The Display feature has four other modes of operation that the user must enter to operate the Directory, Inspect, Local Display, or Program features:

- ***Directory Mode:*** for searching for names/numbers in the system's integrated directory; entered by pressing DIRECTORY.
- ***Inspection Mode:*** for displaying information about call appearances, assigned features, stored numbers, etc; entered by pressing INSPECT.
- ***Local Mode:*** for accessing the built-in clock and timer functions of the display unit; entered by pressing LOCAL at some terminals; default mode at some idle terminals.
- ***Program Mode:*** for supporting the Program feature and storing Repertory Dialing, Flex DSS, and Personal Speed Dialing numbers; entered by moving the PROGRAM switch to its "P" position (except on SLACs) or by dialing access code #4 (at any display set).

General Rules for Normal Mode

- The display tracks, character by character, whatever a user enters from the dial pad buttons. Entries that exceed the capacity of Screen 1 automatically overflow to Screen 2; when this occurs, a continuation symbol "-" appears in position 1 of Screen 2.
  - — If dialed characters form a valid Personal Dial Code (PDC), Data Dial Code (DDC), or Direct Group Calling (DGC) access code, the Display ID from the system's integrated directory is shown.
  - — If dialed characters form a valid feature access code requiring additional input (such as for Account Code Entry), a feature prompt is displayed; the additional input is tracked character by character.
  - — If dialed characters form a valid Personal Speed Dialing code, the number stored under the code is displayed; if no number is stored, NO INFORMATION is displayed.
- The user can alternate between Screen 1 and Screen 2 by pressing the SCROLL button. Pressing SCROLL has no effect if Screen 2 is empty.

- Any information stored by the terminal user (Repertory Dialing numbers, Personal Speed Dialing numbers, etc.) will be displayed when the feature is used. Any information stored by the System Administrator [Automatic Route Selection (ARS) routing information, System Speed Dialing numbers, etc.] is not displayed to the terminal user.

- Displays for redirected/coverage and returning/third-party calls have the following formats:

  — Screen 1 contains either information about the *called* party (usually a station or DGC group) or a feature descriptor that explains why the call has come to this station (for example, PARK RTN or NIGHT SERVICE).

  — Screen 2 *always* contains information about the *calling* party (the person who initiated the call and usually the person on the line when the display station user answers the call).

- Generally, if the System 25 acts on a call without the display terminal user having caused the action, the result is not displayed. For example, if a call is forwarded, there is no indication of forwarding on the caller's display; this also applies to calls that are picked up, sent to coverage, parked, unparked, etc.

  — Exceptions: Conference/bridging indication of the number of parties on a call; exclusion indication when excluded from a call.

- Generally, if the System 25 redirects a call away from a display terminal, the display does not present this information. For example, when a call is picked up by another terminal, the called terminal does not display this action.

  — Exceptions: If a call to a terminal goes to coverage, that terminal's display has a "c" in position 16 of Screen 1; if a call is forwarded to an outside number, the display flashes "F" and the forwarded-to number.

- A call-information display will remain on the screen *unless* one of the following actions occur:

  — The display terminal user presses another call appearance button.

  — The user presses another button that causes a system action and/or has display support.

  — The call disconnects.

  — The user leaves Normal Mode.

  — The user changes switchhook status (from off-hook to on-hook, or vice versa).

  — A call arrives while Automatic Incoming Call Identification is active.

Special Descriptors

The descriptors summarized in Table 2-J appear on displays to provide special information about calls. These symbols consist of upper-case and lower-case letters and other typographical characters. Descriptors on Screen 2 must be interpreted with respect to the contents of Screen 1.

Table 2-J. Special Descriptors

Location	Description	Meaning
Screen 1/Position 1	>	Covered or Redirected Call (more Information on Screen 2)
	}	Returning or Third Party Call (more information on Screen 2)
	F	Call is forwarding from this terminal to an outside location (Screen 2 blank)
	I	Busy-to-Idle Reminder (Screen 2 Blank)
	Q	Queued call (Screen 2 blank, unless more than 13 digits)
	T	Transfer of a call to this terminal is in progress (Screen 2 blank)
Screen 1/ Positions 15 and 16 (If position 15 is blank, check for one of the descriptors in next block.)	^n	Conference or bridging in progress with n active parties (non-SLAC stations only)
Screen 1/Position 16	c	Call has gone from this terminal to coverage (displayed only while the call appearance remains on set)
	0-9, or ! for 10 or more	Number of calls in DGC queue (displayed only to logged-in DGC members)
	1-9, or ! for 10 or more	Number of calls in SLAC queue (displayed only to SLAC attendants)
Screen 2/Position 1	-	Continuation of digit string from Screen 1
Screen 2/Position 16	a	Third-party call setup
	b	Covered or called party is busy
	d	Covered or called party does not answer
	f	Following/forwarded calls
	g	DGC call
	n	Night service call
	p	Picked-up call or returning parked call
	s	Covered party activated Send AH Calls
	u	Call not-signed-in FPDC (SLAC only)

Standard Call Displays

The following basic displays illustrate the arrangement of information on the screen(s) for some of the most common types of calls.

- Origination or Reception of Inside Calls

  When a display set user places or receives an inside call, the other party's extension number appears in positions 1-4 and name (if administered) in positions 6-14. Queue information appears in position 16; this field applies only to SLAC operations and DGC queues, but is reserved on all displays.

  Screen 1 | 307 Martin,H |

- Origination of Outside Calls

  When a display set user places a call to an outside station, the dialed digits appear in positions 1-14 on Screen 1. If the outside number has more than 14 digits, the continuation descriptor "-" followed by the excess digits appear on Screen 2:

  Screen 1 | 9 1 2 1 2 5 5 5 6 0 4 5 1 2 |

  Screen 2 | - 3 4 5 6 7 8 |

  The displays generated for calls placed from System Access, Personal Line, and Pooled Facility buttons do not identify the type button used. The trunk name is not displayed.

- Reception of Outside Calls

  When a display set user receives an outside (trunk) call, the administered Display ID for the trunk appears. For Direct Inward Dialing (DID) and dial-in tie trunks only, the extension number of the called station also appears.

  Example of DID or dial-in tie trunk call:

  Screen 1 | 208 OUTSIDE |

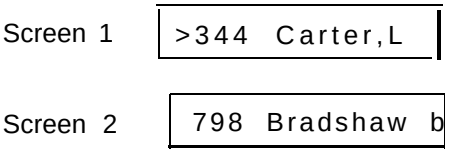
  Example of other trunk call:

  Screen 1 | BRANCH |

- Reception of Coverage or Redirected Calls

  When a display set user receives a call that was originally destined for another station, the redirection descriptor ">" appears in position 1 followed by the covered (or originally called) party number and name. Screen 2 contains the calling party

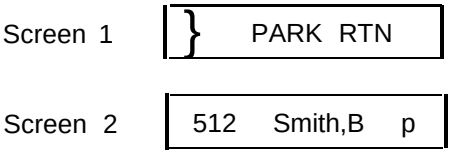
number and name; the call type designator appears in position 16 (here, “b” indicates that the covered party was busy).



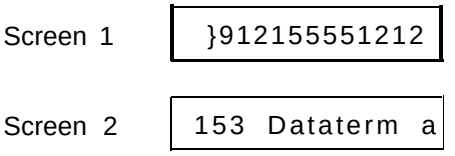
- Reception of Returning or Third-Party Calls

  Screen 1 identifies either where the call is returning from or the destination of the third-party call. Screen 2 presents identification of either the calling party on the returning call or the data terminal that set up the third-party call.

  Example of returning parked call:



  Example of Third-Party Call Setup call at source station:



Special Call Displays

Display enhancements provided by R2V1 apply to the following System 25 features.

- Account Code Entry

  When a user activates the Account Code Entry feature by dialing *0 or pressing ACCT ENTRY, the system displays the prompt ACCT?.

  As the user enters the account code, the digits are displayed to the right of the prompt. If the number of digits exceeds 9, the system automatically scrolls to Screen 2; the continuation character and the remaining digits appear on Screen 2.

  The prompt and digits remain displayed until one of the following occurs:

  — The user enters either “#” or the administered number of code digits.

  — The user restarts the Account Code Entry feature by dialing *0 or pressing ACCT ENTRY again, to correct an erroneous entry.

— The system time-out for Account Code Entry is reached.

— The user selects another button that overwrites the display.

- Busy-To-Idle Reminder

  When a user receives this signal, the display format is the same as when the call was originally placed, except that idle descriptor “1” appears in position 1. Number and name fields are displaced to the right.

- Callback Queuing

  Before this feature is invoked, the display shows the standard format for origination of an inside or outside call. When callback queuing goes into effect for the call, the display updates to CALL QUEUED. If the user cancels queuing, the display is QUEUE CANCELED. If the queuing attempt is denied, QUEUE DENIED appears.

  When a station receives callback, indicating that the called facility is now available, the display shows the same information seen before queuing, except that queue descriptor “Q” appears in position 1, displacing the number and name fields to the right. Once the user answers the callback, “Q” is removed.

- Calls to the Attendant

  The default name associated with PDC “0” is ATTENDANT. A different name can be assigned, if desired.

- Conference/Bridging

  Call descriptor “ ^ ” appears in position 15 of Screen 1 of non-SLAC positions for calls containing more than two active parties; position 16 contains the actual number of conferees or bridgers. The number of conferees is displayed at each terminal in a conference or bridged call and is updated as the status changes.

  Screen 1 | 324 Zimmer,A^4 |

  The “A” and the number of conferees overwrite whatever was in positions 15 and 16 of the current display.

- DGC Queue Field

  A logged-in Direct Group Calling (DGC) group member can view the number of calls waiting to be serviced by the group. The display is continuously updated for all members. A digit, 0 through 9 or “!” for 10 or more, appears in position 16. DGC queue values are not displayed at a SLAC assigned to a DGC group; the attendant’s display always contains the number of calls waiting in the attendant queue.

- FLEX DSS Button Operation

  Operation of a FLEX DSS button that is not programmed with an extension number generates the display NO INFORMATION.

- Forwarding

  Reception of a forwarded call follows the standard format for a redirected call, with the call type descriptor “f” in position 16 of Screen 2.

  A forwarding display station receives abbreviated alert when a call is forwarded to an *outside* number; the display is flashed on Screen 1 only. The new forwarding descriptor “F” appears in position 1, followed by the digits of the outside number.

  Screen 1 | F912325552365 |

- Leave Word Calling (LWC)

  If a user successfully activates LWC, the display shows the called extension number and MSG SENT.

  Screen 1 | 879 MSG SENT |

  If an LWC attempt is not successful, the display shows the called extension and MSG DENIED.

  Screen 1 | 879 MSG DENIED |

- Manual Signaling

  A display set receiving manual signaling from another station has SIG in positions 1-3. The name of the signaling party, if available, or the extension number of the station from which the signal was sent appears in positions 6-14.

  Screen 1 | SIG Borden,L |

  The message is displayed for 5 seconds or until the user selects another button or receives a call. The signaling party has no display for this feature.

- Send All Calls

  When Send All Calls is invoked, the sending station still receives incoming call information. If the calls are being sent to coverage, the proceeding-to-coverage descriptor “c” appears in position 16.

  Screen 1 | 146 Pearson,M c |

- Transfer

  At a station receiving a transferred call, the transfer descriptor “T” is displayed in position 1 before the transfer is completed. The transferring party’s number and name are also shown.

  Screen 1 | T785 Jones, B |

  After the transfer is completed, “T” is removed and the display at the transferred-to station reverts to a standard incoming call format; information about the transferred party is displayed.

  Note: If the transferring station does not have a display and the transferred party is on an outside trunk, the name of the trunk, not the originally dialed digits, is displayed.

Automatic Incoming Call Identification

This subfeature allows a display set user to automatically receive the identification display for a new incoming call while busy on another call. Any existing display is temporarily removed, and the new information is flashed on Screen 1. The display presented by this feature has the same format as the display for an incoming call at an idle station.

The automatic display flashes only once for a given incoming call and appears when the call first rings. If the System Access or Bridged Access button has delayed ringing, the display is delayed also; if the appearance button is administered not to ring, automatic display is suppressed. The automatic display is replaced by another simultaneously incoming call display or by reinstatement of the original active display. If the user wishes to examine the call information again, the Inspection Mode must be entered. When administered for a station, Automatic Incoming Call Identification operates only in Normal Mode, but does not operate while the user is off-hook and dialing digits. Automatic Incoming Call Identification does not involve use of the Inspection Mode. It is administrable on an individual display set basis but is not supported on the SLAC.

Considerations

The Display feature provides valuable call information with a minimum of effort on the part of the voice terminal user.

Interactions

The following features interact with Display Capabilities.

Attendant Console, Direct Trunk: Display capabilities are not supported for this type of console.

Bridging of System Access Buttons: All stations sharing a bridged appearance—that is, the principal station and the bridging station(s)—and having ringing enabled for the appearance will receive standard call information display on incoming calls. After one station answers the call, the bridging station continues to display the call information until the user receives or places another call, the original call ends, or the other station invokes Exclusion.

A bridged appearance user with delayed ringing does not receive incoming call information until ringing starts.

Any station having a bridged appearance can examine the active call information for the appearance by using the Inspect feature, unless Exclusion has been invoked.

If a user at a principal station or a bridging station places a call on a bridged button, outgoing call information is displayed only at the calling station. Other stations sharing the appearance can use the Inspect feature to display the call information.

A user having a bridged appearance can bridge onto a call being originated at a bridged station. In general, the bridging station receives the same call display as the bridged station, but only after dialing has been completed. Conference information is generated and displayed at bridged and bridging stations.

When a user attempts to either bridge onto or Inspect a call, where Exclusion has been invoked by another active party on this call, the screen display shows EXCLUDED.

Callback Queuing: If a user with Automatic Incoming Call ID is off-hook when a callback attempt is made, the display will not flash the callback call's information. However, this information is accessible via the Inspection feature.

Call Waiting: If a display station has Automatic Incoming Call Identification active, call waiting tone is accompanied by an incoming call message flashed on the screen. The user cannot Inspect the message again, because all buttons are busy with other calls. When the user answers a waiting call, the display updates to standard incoming call format.

Conference: When a queued call is added to a conference, the associated displays are modified in only one respect; the Q symbol appears as the first character of the queued call display. When the queued facility becomes available and the call is made, "Q" is removed.

When a nondisplay station originates a trunk call, then conferences the call with an inside display station and drops off, the display shows the trunk name only, not the originally-dialed digits.

Message Waiting Indications: When a display set user dials #90 (or #91) followed by an extension number to light (or extinguish) a Message LED at some station, the dialed digits are displayed. A confirmation of Message LED activation or deactivation is not displayed.

Remote Access: Since remote access calls are all incoming trunk calls, the display at the receiving station has the standard ‘(reception of outside calls” format. The receiving station has no special indication that this is a remote access call.

A display set user who bridges onto a Personal Line Appearance where a remote access call is active will have the display updated for conference status.

Administration Requirements

To implement the Display feature, the following administration items are required.

- Assign display voice terminals.
- Assign LOCAL, SCROLL, and INSPECT buttons on the non-SLAC sets.
- Assign DIRECTORY, CALL, and NEXT buttons to all display sets, including SLACs.
- For each nonattendant display station:
    1. Is this a display station? (yes or no; default = no; change to “yes”).
    2. Enable Automatic Incoming Call Identification? (yes or no; default = no). If this feature is desired, and Step 1 was set to “yes,” no action is necessary; if not desired, set to “no.”
- Set up the system’s integrated directory, giving a Display ID for PDCs, DDCs, DGC groups, DID trunk group, other trunks, and attendant (PDC 0).

Hardware Requirements

To have display capability, a station must be equipped with a Model 7305H04C Multiline Voice Terminal or a Model 7317H01A Multiline Voice Terminal (BIS-34D).

Distinctive Ringing

Description

This feature allows users to distinguish between different types of incoming calls. The system provides the following types of ringing:

- A repeated two-burst tone indicates an outside call or a call extended by the attendant. The two-burst tone pattern is: 0.4 seconds on, 0.2 seconds off, 0.6 seconds on, and 4.0 seconds off.
- A repeated one-burst pattern indicates a call from an internal user. The tone is one second on and three seconds off for multiline voice terminals, and 1.2 seconds on and 4 seconds off for single-line voice terminals.
- A “abbreviated alerting” signal (also called single-ring reminder) indicates to the off-hook user of a multiline voice terminal that a new call is coming into another call appearance button. This type of call rings just once, but the associated status LED continues to flash after the abbreviated alerting stops. The user may place the current call on hold and answer the incoming call if desired.
- A single short beep at a voice terminal equipped with the Hands-Free Answer feature indicates that an incoming inside call has been answered by the terminal. Depending on the status of the terminal’s HFAI controls, the user can talk with the caller without lifting the handset.
- Priority ringing is a repeated pattern of two short rings followed by one long ring. It indicates (1) that a data terminal has used the Third-Party Call Setup feature to originate a voice call from the voice terminal where this ringing is heard, or (2) that a queued-for facility is now available and the user can go off hook for the call to be completed.

Considerations

Distinctive Ringing enables a user to handle each call in an appropriate manner.

Abbreviated alerting notifies the busy called party of an incoming call without the annoying distraction of continued ringing.

Distinctive ringing is not available at Extended Stations; all incoming calls are signaled by standard one-burst ringing, repeated.

Interactions

The following feature interacts with Distinctive Ringing.

Coverage: Covering stations receive distinctive ringing, depending on the origin of the call receiving coverage.

End-To-End Signaling

Description

This feature allows multiline voice terminals to send touch-tone (DTMF) signals over the DDD network and allows single-line and multiline users to send touch-tones over dial pulse trunks.

The 7300H series voice terminals do not generate touch-tones when a dial pad button is pressed. The End-to-End Signaling feature provides for the conversion of signals generated by these terminals to touch-tones.

Dialed numbers from multiline voice terminals are toned out for a default duration of 60 ms followed by 60 ms of silence (administrable). Dialed numbers to single-line voice ports are toned out for a default duration of 60 ms followed by 60 ms of silence (administrable).

When using dial pulse trunks, End-to-End signaling is invoked by dialing "#" after the last digit of the *called* number or waiting for about 10 seconds after dialing the last digit (see the Interdigit Timeouts feature description). All subsequent dial pad button presses generate touch-tones on the outside line.

Considerations

End-to-End Signaling permits stations to access network services that require touch-tone signals.

Interactions

The following features interact with End-To-End Signaling.

Command Mode And Data Terminal Dialing: Occasionally it is necessary to send additional tones to the remote endpoint after a data connection has been established. A mark character "$" is embedded in the dialing sequence to indicate to call processing that additional tones must be sent prior to insertion of a conversion resource (pooled modem) into the connection. The mark character "$" is used to indicate that all the following digits are for end-to-end signaling. This character is used to mark the boundary between the digits dialed to reach the distant endpoint and the digits used by the distant endpoint after it answers.

Repertory Dialing: Repertory Dialing can be programmed on the 7300H series voice terminals. End-to-End Signaling works properly with this feature.

Speed Dialing: #8 must be stored to start End-to-End Signaling.

Virtual Facilities: #8 must be stored to start End-to-End Signaling.

Exclusion

Description

This feature allows multiline voice terminal users to keep other users with appearances of the same Personal Line from listening in on or interrupting their calls. It can also be used in a Principal Station/Bridging Station arrangement by either party to exclude other inside stations from a private call. Exclusion allows users to exclude the attendant and other stations from an existing or held call, or to drop other System 25 users from a call.

The EXCLUSION button status and l-use LEDs are lighted steadily when the feature is invoked. When an excluded call is placed on hold, the EXCLUSION button's l-Use LED goes dark and the status LED winks with the LED of the held line.

Exclusion can be applied to only one call at a time. Once Exclusion is invoked on a call it will remain active until the user either presses the button a second time or disconnects the call.

Considerations

Exclusion allows the sharing of a Personal Line or a bridged System Access (SA) appearance by several users while retaining privacy for each one.

Pressing the EXCLUSION button at any time during a call, regardless of how the call was originated, drops all other inside stations and tones. An inside party can be included on a private call by pressing EXCLUSION first and then adding the inside party.

Interactions

The following features interact with Exclusion.

Automatic Intercom: Any attempt to activate Exclusion while active on an Automatic Intercom call will drop the other party.

Bridging of System Access Buttons: If a principal or bridging station presses the EXCLUSION button during a call, all other internal stations on the call will be dropped. In addition, Exclusion will prevent any other internal station from bridging onto the call.

Callback Queuing: Pressing the EXCLUSION button does not drop a queued call. The EXCLUSION button's status LED tracks the status LED of the associated call button. For example, on a callback attempt, the EXCLUSION LED will also change from winking to flashing. If the EXCLUSION button is tracking a conference on hold, it will stay winking with the rest of the conference. When the callback attempt is answered, the EXCLUSION LED lights steadily to track all the conference buttons.

Conference: When Exclusion is invoked, all other inside parties will be dropped. If a private conference including inside parties is desired, the user should activate Exclusion first and then set up the conference.

Coverage: If a coverage receiver invokes Exclusion after answering a coverage call, all other terminals (including the attendant and the covered station) are excluded. The covered user cannot enter the call until Exclusion is pressed a second time by the covering user.

Display: When a display station attempts to enter a personal line or bridged appearance that has the Exclusion feature in effect, EXCLUDED is displayed.

Hold: A call can be placed on hold after Exclusion is invoked. The l-Use LED will go dark; the status LED of the line appearance button and the Exclusion button will wink.

Administration Requirements

Voice Terminal Port:

- Assign EXCLUSION button.

Expert Mode

Description

Expert Mode is an enhancement to the Command Mode feature that provides an alternative method of performing the full range of Command Mode functions. By eliminating the display of menus and allowing multiple commands to be entered on a single line, Expert Mode lends itself to computer-driven instructions. Individual users who are very familiar with Command Mode operations may also find it useful.

When Expert Mode is activated, a system administrable prompt is displayed that can consist of up to nine characters (the quote character and RETURN are not allowed). **Command:** is the system default prompt. As with dialing in Command Mode, the ASCII characters backspace (BS or CTRL-H) or underscore (–) may be used to cancel a previously entered character. *When in Expert Mode, each line must be terminated with a keyboard RETURN.*

Users of Expert Mode **must** follow the exact tree structure of Command Mode (both up and down the menu tree) as shown in Figure 2-55. However, instead of moving one level at a time, Expert Mode allows the user to move up or down several menu levels at once. This can be accomplished by entering, on a single command line, the capitalized letters that define the sequence of menu selections desired. For example, to change data port parity from the tree's entry level, the user types OCPE and presses RETURN. This requests that parity be set to "even", but does NOT enable the change. To enable this change (see Figure 2-55), the user must now type XE and press RETURN.

Activating Expert Mode

A user can move back and forth between Command Mode and Expert Mode by typing ''!" (exclamation mark). For ports on a Data Line circuit pack (Data Line Card, DLC), either Command Mode or Expert Mode is presented at the start of a new session, depending upon the port's setting at the termination of the previous session. Thus, if a data session ends in Expert Mode, the next session will begin in Expert Mode. However, calls from an AT&T STARLAN NETWORK to System 25 will *always* begin a new session in Command Mode.

An alternative command, ">", can be used to *guarantee* entry into Expert Mode. Conversely, guaranteed entry into Command Mode can be accomplished with the command ">!" followed by RETURN. These commands are especially useful for computer-driven DLC endpoints that might otherwise have difficulty detecting whether a new session had been started in Expert Mode or Command Mode.

Making a Data Call

To make a data call from the entry level (see Figure 2-55), the user enters "D" following the system prompt and then the data endpoint number. For example:

Command: D9,5553822

"Command:" on the above line is the default system prompt while in Expert Mode. The user enters all data following the prompt.

If the user enters “D” and then a RETURN, the system will prompt for the data endpoint number as follows:

Command: D
DIAL:

The user must then enter the digits required to complete the call.

Activating the Third-Party Call Setup Feature

The following provides an abbreviated method of using the Third-Party Call Setup feature while in the Expert Mode. A complete description of this feature is provided later in this manual.

To activate the Third-Party Call Setup feature and place a call, the user enters numbers using the following format:

Command: V{ Destination} F {Source}

The V on the above command line provides access to **<Voice call>** from the Command Mode entry level menu. The balance of the dialed number is composed of *destination* and *source* numbers, as described in the Third-Party Call Setup feature description.

User Changeable Options

Refer to the User Changeable Options feature (discussed later) for a detailed description of the feature. The menus selected in the following discussion are shown in Figure 2-55.

To view the current Options Table (starting at the Command Mode entry level), the user simply enters “OV” following the system prompt, as follows:

Command: OV

To change the current Options Table (starting at the entry level), the user enters “OC” as follows:

Command: OC

Entering “OC” places the user at the Change Options level. At this point the user may change options by entering the appropriate letter to indicate the required option (S for Speed, P for Parity, M for Mismatch, etc.) followed by the desired setting(s). Only one Option is allowed per line. If more than one setting is selected for an Option that can only accept one setting, call processing recognizes only the last entry.

Examples:

Command: S +1200-300+4800	*Add 7200 and 4800 baud to the available speeds, remove 300 baud*
Command: PE	*Change Parity to Even*
Command: MY	*Change Mismatch to Yes*

If the user enters an invalid Option or setting, the system responds with INVALID OPTION and the entry is ignored.

Once all changes have been entered, the user enters XE to enable the options.

Considerations

Expert Mode is primarily for use by computer-driven endpoints that can store command sequences for automated use. However, a user experienced in accessing Command Mode menus may find Expert Mode to be a faster alternative when operating at slower speeds, since the time required to display each menu and to input separate commands is essentially deleted.

Interactions

The following feature interacts with Expert Mode.

Command Mode: Refer to the Command Mode feature description for a detailed description of Command Mode and of the various menu items.

Administration Requirements

The data port associated with a data terminal can be administered to allow the user to change options when in Command or Expert Mode. Otherwise, the user may view the current options but not change them.

The default prompt for Expert Mode **(Command:)** may be changed via system administration.

Extended Stations

Description

Allows single-line voice terminals to be located at distances from 2000 to 17500 feet from the system cabinets.

Extended stations have the same feature capability as other voice terminals. These stations count as an outside party on conference calls.

Transmit and receive levels are increased at extended stations for conferencing.

Considerations

A single-line voice terminal must be administered as an extended station before this feature is activated.

Extended stations will always receive standard (that is, single) ring for calls; System 25 will not send distinctive ringing.

Interactions

The following feature interacts with Extended Stations.

Conference: An Extended Station counts as one of the two outside parties allowed on conference calls.

Administration Requirements

Single-Line Voice Terminal Port:

- Assign port on Analog Line (TN742) CP.
- Make This An Extended Station (yes or no; default = no).

Hardware Requirements

The Extended Station must be a single-line voice terminal. It requires a port on a TN742 Analog Line CP.

External Alerts

Description

External Alerts provide standard station ringing at locations away from the called stations. This feature can be used to activate an external alerting device such as a bell.

External Alerts supports the Trunk-Answer-from-Any-Station (TAAS) form of Night Service.

The feature can be used in conjunction with voice terminals located in noisy environments and large areas such as warehouses, etc. The alerting device is activated whenever the associated station is alerted.

A Supplemental Alert Adapter installed on a hybrid station allows the terminal user to transfer incoming ringing to an alerting device located in some remote area. When the user goes to the area, the alerting device rings for incoming calls to the user's normal station.

Considerations

External Alerting enhances user ability to recognize incoming calls. Noisy environments, large areas, and outside locations are candidates for external alerting devices.

Interactions

The following features interact with External Alerts.

Manual Signaling: Manual Signaling will not activate an external alerting device.

Night Service: When the system is in Trunk-Answer-f rem-Any-Station (TAAS) Night Service mode, an incoming attendant-seeking call will activate the Night Service alerting device.

Power Failure Transfer: When the system is in the power failure transfer mode, the external alerting devices are disabled.

Administration Requirements

Special Feature Port:

- An external alert operating as the endpoint device on a station line requires a port assignment on a ZTN78 Tip Ring Line or TN742 Analog Line CP. (Specify special feature port type = 253.) An external alerting device controlled from a Supplemental Alert Adapter operates on the same line as the associated terminal and requires no additional port assignment.
- Specify the PDC of the associated station (or 0 if alert is used with TAAS Night Service).

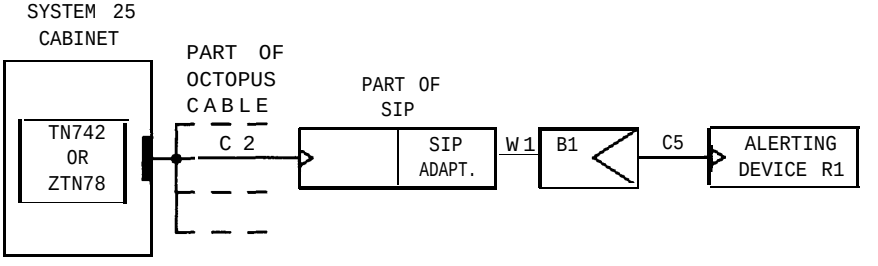
Hardware Requirements

Order line-activated alerting devices (e.g., bells) as required.

An alerting device operating on a line separate from a terminal requires a port interface on a ZTN78 or TN742. Refer to Figure 2-19 for connection information.

Order the Supplemental Alert Adapter (PEC 2301-SAA) for controlling a remote alerting device. A Supplemental Alert Adapter is installed in the line between the port CP and the user's MERLIN System terminal. The line requires a ZTN79 ATL Line CP. Figure 2-20 contains connection details.

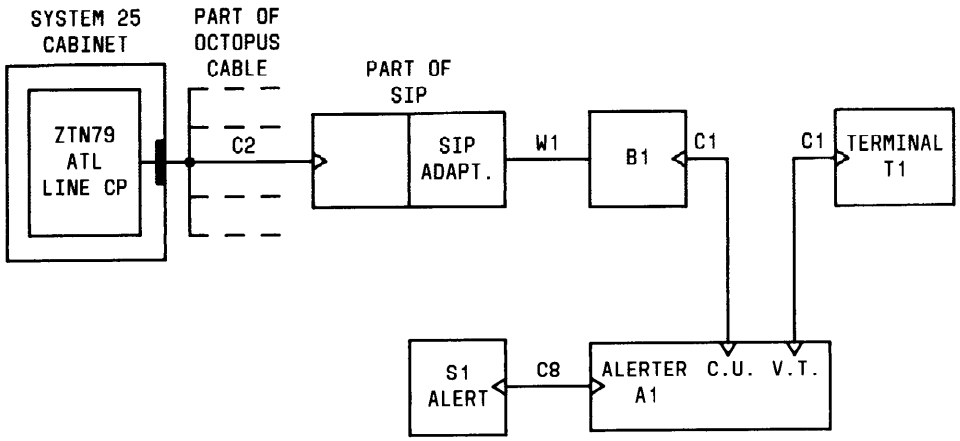
Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters are provided under the heading "Connectivity" in Section 4.



LEGEND:

TN742 - ANALOG LINE CP
ZTN78 - TIP RING CP
81 - TYPICAL- 103A CONNECTING BLOCK*
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C5 - MODULAR CORD (D4BU-87)
R1 - E1CM-50 RINGER OR EQUIVALENT PEC-31019
W1 - 4-PAIR INSIDE WIRING CABLE*
* - FURNISHED BY INSTALLER

Figure 2-19. External Alert Connections



LEGEND:
- A1 - SUPPLEMENTAL ALERT ADAPTER –PEC 2301-SAA
- B1 - TYPICAL – 103A CONNECTING BLOCK*
- C1 – MODULAR CORD (D8W-87)
- C2 - OCTOPUS CABLE (WP90780)
- C8 – MODULAR CORD (D4BU-87)
- S1 – EXTERNAL ALERT
- T1 – HYBRID TYPE TERMINAL
- W1 - 4-PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

Figure 2-20. Supplemental Alert Adapter Connections

Following

Description

This feature allows users who are away from their own voice terminals to receive their calls at other inside voice terminals. Following is functionally equivalent to internal Forwarding.

In order to have their calls follow them, users sign in their PDCs at the terminals where they will be located temporarily. A call coming into the "home" terminal is redirected to the terminal where the PDC is signed in (the "away" terminal). Signing in a PDC at a terminal automatically signs the PDC out at any other terminal. PDCs always have an associated home terminal. If a PDC is not signed in anywhere, calls to the PDC will ring at the home terminal.

Signing in a FPDC automatically signs the FPDC out at any other voice terminal. However, signing out a FPDC does not sign in the FPDC at another terminal. Calls to FPDCs not signed in at any terminal may be directed to the attendant (administrable and strongly recommended).

To sign in a PDC, the user goes off-hook at the away terminal, enters "**" and then the home PDC twice. The PDC sign-out procedure is similar; after going off-hook and receiving dial tone, the user enters "**" followed by the home PDC and then by a "0." A "**O" entered at a voice terminal will sign out all PDCs and FPDCs signed in at that voice terminal, except for the PDC normally assigned to it. Confirmation Tone is returned to a user who correctly completes one of these procedures.

The call types or features listed below are voice terminal oriented (associated with stations rather than PDCs) and do not follow a user who signs in at another terminal.

- Automatic intercom Calls
- Callback calls when a queued-for facility becomes available
- Calls ringing on Bridged Access (BA) buttons
- Coverage calls
- Directed Night Service calls
- DGC Group Calls
- Manual Signaling
- Message Waiting indications
- Outward/Toll Restriction
- Personal Line Calls
- Returning calls

Calls Placed to A PDC:

Calls to a PDC that is not signed in at an "away" terminal are directed to the home terminal and receive that terminal's normal hunting or coverage treatment.

If a PDC is signed in at an away terminal, calls placed to the PDC fall into one of the following categories:

- The general case—calls placed from terminals other than the away terminal
- A special case—call placed from the away terminal.

(1) The General Case: The call will first be directed to the away voice terminal. Ringing will occur at the away terminal if it is an on-hook single-line voice terminal or if it is a multiline voice terminal with an idle System Access button.

A call unanswered at the away terminal will be directed back to the home terminal unless one of the following busy conditions exists at the home terminal: (1) it is a multiline terminal with all System Access buttons busy and with no idle coverage receiver or (2) it is a single-line voice terminal that is off-hook, has no idle coverage receiver, and has no idle hunt-to station. While either of these busy conditions exists at the home terminal, the call will not return to the home terminal. Instead, it will continue to ring at the away terminal until answered or timed out.

If either of the above two busy conditions ends at the home terminal while the call is waiting at the away terminal, the call will be directed back to the home terminal. If the call is sent back to the home terminal, it can be answered or it can receive the terminal's hunting or coverage treatment. The coverage treatment given to calls that are returning from an away terminal differs in two respects from the treatment provided to calls initially directed to the home terminal. First, the home terminal and its coverage station receive ringing simultaneously, rather than having the coverage ringing delayed. Second, coverage terminals will ring for calls returning to the home stations even if the coverage ring options of the home terminal are "no ring. " Both of these coverage modifications expedite the answering of calls that are returning to the home station from an away station.

Once the call is directed back to the home terminal, it is removed from the away terminal. This is true even if the away terminal was busy but subsequently became idle after the call was sent back to the home terminal.

(2) A Special Case: A call to a PDC placed from the same station where it is signed in will be directed to the PDC's home terminal.

Calls Placed to an FPDC:

When a valid FPDC is dialed, the call will be directed to the terminal where the FPDC is signed in and will be provided the coverage treatment administered for that terminal. If the FPDC is not signed in anywhere and if the attendant position is administered to handle these calls, then the call will be directed to the attendant position. However, if the FPDC call was placed from the attendant position, then it will not be redirected to the attendant but will

instead be provided Reorder Tone. Finally, if the FPDC is not signed in, and if the attendant position is not administered to handle these calls, then the calling party will receive Reorder Tone.

For non-DID calls, if an invalid FPDC is dialed, then the calling party will receive Reorder Tone. If a DID call does not match any assigned number in the dialing plan, it will be directed to the attendant or to Reorder Tone, as administered.

Considerations

Following provides maximum flexibility to system users who are away from their voice terminals. In addition, visitors can receive calls by signing in an assigned FPDC.

For more information, see the “Personal Dial Code (PDC)” feature description.

Interactions

The following features interact with Following.

Bridging of System Access Buttons: Sign-in and sign-out procedures can be performed at the destination station on either a System Access (SA) button or a Bridged Access (BA) button. However, since Following calls always arrive on SA buttons, the destination station must have at least one SA button.

Following calls arriving at a principal SA button are accessible at BA buttons on the bridging station.

Callback Queuing: Calls that follow are queued on the busy “away” station, not the “home” station.

Callback attempts to the originator do not follow.

Coverage: Calls to a signed-in FPDC receive the coverage of that terminal. Unanswered calls to a PDC at an away terminal return to the home terminal and receive the home terminal’s coverage treatment; they do not receive the away terminal’s coverage.

Forwarding: Either Following or Forwarding, but not both, can be active at a given time for a particular PDC. Activation of one feature while the other is in effect overrides the other feature.

Remote Access: Remote Access callers cannot use Following. However, Remote Access callers can activate Forwarding; see the “Forwarding” feature description for information on this capability.

Forwarding

Description

This feature allows users to direct their incoming calls to another (forwarded-to) voice terminal where they will be located temporarily. Calls can be forwarded to inside stations or to locations outside System 25. Figure 2-21 is a simplified block diagram of this feature.

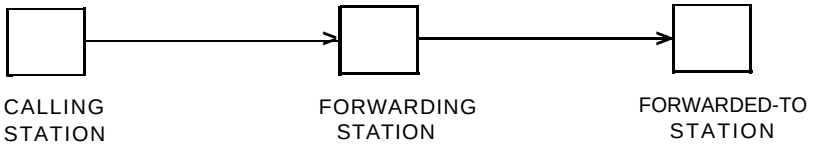


Figure 2-21. Stages of Call Forwarding

Forwarding is similar to the Following feature. Both features enable users to answer their calls at another terminal. The basic differences between these features are as follows:

- Users activate Forwarding at their own stations; the forwarded-to station can be inside or outside System 25.
- Users activate Following at the "away" station, which must be inside System 25.

The procedures for activating the two features are different also. Refer to the "Following" description for complete coverage of that feature.

Forwarding Calls to Stations Inside System 25

The user activates Forwarding to an inside station by dialing feature access code #70 and then the forwarded-to extension number. If the activation is successful, the user receives confirmation tone; if the attempt fails, the user hears reorder tone.

The user who forwarded calls to an inside station can deactivate Forwarding in either of these ways:

- By signing in at the user's own (the forwarding) station after returning there; this procedure consists of dialing "**" and then the user's PDC twice.
- By signing out at the forwarded-to station before returning to the forwarding station; this procedure consists of dialing "**" followed by the user's own PDC and then "0".

Calls forwarded to an inside station cannot forward again from the forwarded-to station.

A call unanswered at a forwarded-to inside station returns to the forwarding station after the administered number of rings; if the forwarding station has Coverage, the call redirects. A call forwarded to a busy inside station rings the forwarding station immediately; the call returns ringback to the caller unless the caller has automatic Callback Queuing or Call Waiting is administered at the forwarded-to station.

Forwarding Calls to Locations Outside System 25

The user activates Forwarding to an outside station in one of the following ways:

- By dialing feature access code #70, the single-digit ARS access code, and the forward-to number, then hanging up after hearing confirmation tone.
- By dialing feature access code #70, a pooled facility (trunk) access code, and the forward-to number, then hanging up after hearing confirmation tone.

The forward-to numbers for outside calls must be dialed in one of the following ways:

- 7 digits, " 1" plus 7 digits, 10 digits, or "1" plus 10 digits.
- A Speed Dialing code, such as #20.
- By adding # at the end of a code of more than 4 digits and less than 10 digits.
- By letting the system time-out after entering a code of more than 4 digits and less than 10 digits.

If the forwarding activation attempt is unsuccessful, the user hears reorder tone.

Forwarding to an outside station is deactivated by signing in at the user's own (the forwarding) station inside System 25. The procedure consists of dialing "**" and then the user's PDC twice. No procedure exists for signing out at an outside forwarded-to station.

No matter which ring option ("no ring, " "immediate ring, " or "delayed ring") is administered for the forwarding station's SA buttons, the station will receive an abbreviated alert (single-ring reminder) when a call is forwarded outside.

Forwarding to an outside location is enabled or disabled on a per-station basis through system administration. In addition, forwarding calls to outside stations is limited by any calling restrictions administered for the forwarding station. An outward-restricted station, for example, cannot forward calls out of the system. Toll and facility access restrictions can also prevent call forwarding.

A call forwarded to an outside station that is busy or does not answer is treated like any network call; the caller receives busy tone or ringback. The call will not return to the forwarding station. In certain circumstances, incoming *outside* calls that are forwarded to outside locations may encounter an unusual sequence of tones. For example, the caller may hear ringing, a pause, then busy tone.

Remote Access Forwarding

Forwarding to an outside number from a System 25 station can be activated by a Remote Access user calling into the system on a dedicated Remote Access trunk or on a shared trunk while Night Service is in effect; the caller must use a barrier code. Remote Access Forwarding to other inside stations is not allowed.

After dialing the Remote Access trunk and the barrier code, the caller receives second dial tone. The caller then enters access code #70 and the PDC of the forwarding station; confirmation tone followed by silence is returned if a valid PDC was dialed. Finally, the remote caller dials the outside forward-to number and hears confirmation tone.

To cancel Remote Access Forwarding, the remote caller repeats the activation procedure but substitutes “0” for the outside forward-to number.

Considerations

Forwarding helps System 25 users avoid missing important calls while they are absent from their “home” terminals. It complements coverage features by allowing users to answer their own calls remotely rather than have other users take their messages. Calls can be forwarded from all types of voice terminals except rotary-dial sets.

A forwarded-to multiline voice terminal must have at least one System Access (SA) button. If a user attempts to forward calls to a station without an SA button, reorder tone is returned, and the attempt is blocked.

In general, the only calls that forward from a terminal are internal calls, transferred or attendant-extended outside calls, Remote Access trunk calls, and DID calls. At multiline terminals, such calls ring at SA buttons.

The following call types or features are station-oriented (rather than PDC-oriented) and do not forward:

- Automatic Intercom calls
- Callback calls when a queued-for facility becomes available
- Calls ringing on Bridged Access (BA) buttons
- Coverage calls
- Directed Night Service calls
- DGC Group calls
- Manual Signaling
- Message Waiting indications
- Outward/Toll Restriction
- Personal Line calls
- Returning calls.

An attendant’s PDC can serve as a forward-to point for other stations. Calls placed to the attendant’s PDC can be forwarded by the attendant.

A given station can receive forwarded calls from any number of other stations.

Note: When incoming trunk calls are forwarded to outside locations, severe attenuation of the voice signal may occur.

Interactions

The following features interact with Forwarding.

Account Code Entry, Forced: Stations with this feature administered for all calls cannot forward calls to any outside numbers. Stations with this feature administered for "dial 0 or 1" calls can forward calls to any outside number except for "dial 0 or 1" numbers.

Attendant Call Extending: Calls extended by an attendant to a forwarding station will be given normal Forwarding treatment.

Bridging of System Access Buttons: Since forwarding is a station-oriented feature, it can be activated and deactivated for a forwarding principal station only at a System Access button on that station. If forwarding is activated at a Bridged Access button on a bridging station, it affects calls to that station only.

Callback Queuing: Calls that forward are queued for the busy "away" station, not the "home" station.

Callback attempts to the originator do not forward.

Conference: If one of the called parties for a conference is a forwarding station, its forwarded-to station will be the conference facility.

If a conference call is transferred to a forwarding station, it will be given normal Forwarding treatment.

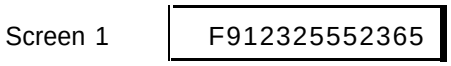
Coverage: When a station has both Coverage and Forwarding in effect, calls are routed first to the forwarded-to station. If not answered there within an administered number of rings, calls ring at the forwarding and coverage stations and stop ringing at the forwarded-to station.

When forwarding to an outside number, coverage may only occur in one case: the forwarding had been activated using a trunk group's facility access code (not ARS), the forwarding cannot be completed because the trunk group is busy, and the forwarding station is not busy. In this case, the call will ring at the forwarding station and its coverage stations.

Direct Station Selection (DSS): DSS or FLEX DSS cannot be used when forwarding calls (that is, dialing #70 and pressing a DSS or FLEX DSS button for the forwarded-to station is not a valid procedure).

Display: Reception of a forwarded call follows the standard format for a redirected call, with the call type descriptor “f” in position 16 of Screen 2.

A forwarding display station receives abbreviated alert when a call is forwarded to an *outside* number; the display is flashed on Screen 1 only. The new forwarding descriptor “F” appears in position 1, followed by the digits of the outside number.



Following: Either Following or Forwarding, but not both, can be active at a given time for a particular PDC. Activation of one feature while the other is in effect overrides the other feature.

Remote Access: Remote Access calls to a System 25 station that has Forwarding activated will forward like any other incoming calls to the station.

Remote Access callers can activate Forwarding to outside numbers at System 25 stations. Refer to “Remote Access Forwarding” earlier in this feature description.

Repertory Dialing: The forwarding activation and deactivation sequences (or portions of them) can be stored on REP DIAL buttons.

Send All Calls: Forwarding supersedes Send All Calls. A call forwarded from a station with Send All Calls activated will not go to Coverage or to bridging stations unless the call is not answered at the forwarded-to station and returns. After returning, the call routes according to the Send All Calls feature.

Station Hunting: Calls forwarded to a station in a hunt group will hunt and ring an idle station if the forwarded-to station is busy. If all members of the group are busy and the forwarded-to station has Call Waiting, the caller hears special ringback until the forwarded-to station becomes available to answer the call.

A call to a forwarding station in a hunt group will first ring at the forwarded-to station. After an administered number of rings, the call returns to the hunt group; if all members of the hunt group are busy, the call continues to ring at the forwarded-to station until a hunt group member becomes available.

Station Message Detail Recording (SMDR): When a call is successfully forwarded to an outside number, the call record will contain the forwarding station and forwarded-to station numbers if Forwarding was activated by an inside station. If Forwarding was activated remotely, the SMDR call record will contain the incoming trunk number, the PDC of the forwarding station, and the barrier code number. For more details, see the “Station Message Detail Recording” feature description.

Transfer: Calls transferred by TRANSFER button operation to a forwarding station will be given normal Forwarding treatment.

Trunk Groups, Loop Start: If the System 25 uses Loop Start trunks, calls can be forwarded to remote locations *on/y* if Trunk-to-Trunk Transfer has been administered for Loop Start trunks.

Administration Requirements

Voice Terminal Port:

- Allow this station to forward calls to outside Locations? (yes or no; default = no).

Hands-Free Answer on Intercom (HFAI)

Description

This feature allows the following voice terminals to provide hands-free answer service on eligible incoming calls; each terminal must have Automatic Answer (AUTO ANS) assigned to a flexible button.

- BIS (7305H03B, 7305H04C, 7313H01A, 7314H01A, 7316H01A, and 7317H01A) and HFAI (7309 HOI A); these sets provide full service without requiring adjuncts.
- 10-Button (7303H01B), 34-Button (7305H01B), and 34-Button Deluxe (7305H02B) equipped with a Hands-Free Unit (HFU—a S102A Speakerphone); these arrangements provide full HFAI service.
- 5-Button (7302H01C) and the 10- and 34-Button sets listed above, not equipped with an HFU; these arrangements allow callers to ‘(voice announce” their calls, but the terminal user must use the handset to reply.

Calls Eligible for Hands-Free Service:

- Inside calls (that is, calls from one System 25 set to another System 25 set using a System Access, Loop, DSS, or Auto Intercom button).
- Calls transferred from another System 25 set using the Transfer feature. The *transferring* station may pass both inside and outside calls in this way. Note that calls *transferred* by the attendant are indistinguishable from calls transferred by any other station.

Calls Not Eligible for Hands-Free Service:

- Incoming trunk calls (Personal Line, DID, DGC).
- Calls extended by an attendant.

BIS and HFAI Voice Terminals

LEDs next to the AUTO ANS button and the HFAI/Mic (HFAI set) or HFAI (BIS set) button indicate whether the HFAI feature is enabled. The LEDs are turned on and off by pressing the adjacent buttons. When both the AUTO ANS and HFAI LEDs are on, the set will auto-answer eligible calls.

The HFAI LED will wink (on HFAI sets) or light steadily (BIS sets) during HFAI calls.

The set’s response to HFA1-eligible calls depends on the status of the HFAI and AUTO ANS buttons and LEDs, as follows:

- If both HFAI and AUTO-ANS LEDs are on:

— The set generates a tone burst over its speaker to indicate an incoming call.

— The parties may converse. The called party can speak in a normal voice toward the set. No other action by the *called* party is required.

— During the call, the called party can press the HFAI/Mic or MICROPHONE button to mute the microphone temporarily and prevent the caller from hearing. Pressing the button again turns the microphone on again.

— The HFAI/BIS user may press the SPEAKER (HFAI set) or the SPEAKERPHONE (BIS set) button to end the call. If the calling party hangs up first, this is not necessary.

- If only the AUTO ANS LED is on:

  — The set generates a tone burst over its speaker to indicate an incoming call.

  — The set's speaker turns on and the set "answers" the call.

  — Call setup is complete. However, the called party can hear, but not respond to, the calling party. To respond, the user must lift the handset or press the HFAI/Mic button on an HFAI set or press the MICROPHONE button on a BIS set.

  — The HFAI/BIS user may press the SPEAKER (HFAI set) or the SPEAKERPHONE (BIS set) button to end the call. If the calling party hangs up first, this is not necessary.

- If only the HFAI LED (or neither LED) is on:

  — The HFAI feature is disabled. The call answering procedure is the same as for a standard MERLIN System set.

If, during an HFAI call, the user decides to pick up the handset, the HFAI/Mic or HFAI LED will turn off. On an HFAI set, the user is not permitted to revert to hands-free operation. (Pressing the HFAI button while using the handset will simply disable the HFAI feature for subsequent calls.) A BIS set user may transfer a call from the handset to the speakerphone by pressing the SPEAKERPHONE button and hanging up.

Voice Terminals with Speakerphone or Headset Adjuncts

These sets do not have an HFAI button. To turn on the HFAI feature the user simply presses the AUTO ANS button; the green status LED lights.

After HFAI is activated, operation is exactly the same as for the BIS set except that the SPEAKERPHONE and MICROPHONE buttons and LEDs are on the HFU.

Note, that the 502B Headset Adapter is required for HFAI operation with a headset (as is usually desired in ACD operation). The 502A Headset Adapter does not support HFAI operation.

Voice Terminals without Speakerphone or Headset Adapters

The HFAI feature is activated by pressing the AUTO ANS button. A beep signal announces an incoming call and the SPEAKER LED lights. A one-way talking link is established from the caller to the terminal; the user can hear the caller but cannot converse. Lifting the handset connects the user to the caller.

Considerations

The user of a HFAI equipped station should always deactivate the HFAI feature when leaving the work area. If this is not done, incoming calls will be unintentionally “answered.”

Interactions

The following features interact with Hands-Free Answer on Intercom,

Bridging of System Access Buttons: If a station has HFAI activated, internal calls arriving at this station on a System Access button will auto-answer. However, calls arriving at this station on a Bridged Access button will ring according to the administered ring option and will not auto-answer.

Coverage: When the HFAI feature is enabled at a set, calls eligible for HFAI service will not receive coverage because the set will answer them whether the user is present or not. However, if the attendant uses the Attendant Message Waiting feature to turn on the Message indicator at the set, the HFAI feature will be disabled (the AUTO ANS LED turns off), allowing subsequent calls to receive coverage.

Send All Calls: Activating Send All Calls will disable the HFAI feature (the AUTO ANS LED turns off).

Administration Requirements

Voice Terminal Port:

- Assign AUTO ANS button.

A 22- or 34-button built-in speakerphone (BIS) voice terminal (with or without display) should be translated as Type 308 *only* if a headset adapter will be used with it. Otherwise, such a terminal should be translated as Type 305.

Hardware Requirements

This feature requires one of the voice terminals or combinations of terminal and adjuncts listed in the Description.

Headset Adapter Adjunct

Description

The headset adapter adjunct is an interface device for connecting a headset to an associated voice terminal. A headset plugged into the adapter is activated by switches on the adapter. The terminal operator has the choice of using either the handset or the headset for handling calls. Turning the headset on and off is equivalent to lifting and hanging up the handset.

Considerations

Use of a headset allows a voice terminal operator to carry on conversations with both hands free for writing, typing, etc. It is valuable adjunct for high traffic positions such as attendant consoles.

Use of a headset does not affect normal voice terminal operations in any way.

Interactions

The following feature interacts with Headset Adapter Adjunct.

Speakerphone Adjunct: A voice terminal cannot have both a headset and a speakerphone. These adjuncts plug into the same jack on the voice terminal.

Administration Requirements

A 22- or 34-button built-in speakerphone (BIS) voice terminal (with or without display) should be translated as Type 308 *only* if a headset adapter will be used with it. Otherwise, such a terminal should be translated as Type 305.

Hardware Requirements

500A/502A/502B Headset Adapters:

The 500A adapter (Figure 2-22) is designed for use with the 12-Button (7203M) MET voice terminal. The 502A and 502B adapters, which are identical in appearance, are designed for use with MERLIN System (7300H Series) terminals (with the exception of the 5-Button and HFAI sets). Most standard commercial headsets can be used with the adapters. The 502B adapter must be ordered if the user requires HFAI operation on the headset (i.e., typical ACD/CMS operation).

Each adapter has an “ON/QUIET” button, an “OFF” button, a green indicator lamp, a jack for a single headset, and a modular keyed 8-wire jack. Each adapter is equipped with an 18-inch connecting cord. Optional cords are available in lengths of 4 and 14 feet.

The 500A Headset Adapter is powered locally by a 2012D Transformer, which plugs into a 115V ac receptacle. Power from the transformer is applied to the voice terminal mounting cord via a 400B2 adapter at the wall jack and conducted to the 500A on its connecting cord. Refer to "Voice Terminal Adjunct Power Supplies" in Section 4 for additional information.

The 502A and 502b Headset Adapters do not require supplemental power, except when used with a 34-Button Deluxe, 22-Button BIS, 34-Button BIS, or BIS with Display voice terminal, or when located more than 200 feet from the switch.

Detailed headset adapter connection information is provided in the following figures:

- Figure 2-23— Typical Headset Adapter to 7300H Series Voice Terminal Connections Not Requiring Auxiliary Power
- Figure 2-24—Typical Headset Adapter to 7300H Series Voice Terminals Connections Requiring Auxiliary Power
- Figure 2-25—Typical Headset Adapter Connections for 12-Button MET Sets

MET Headset Adapter:

Use of a headset with a 10-Button MET voice terminal requires a JS0180-3A Headset Adapter (18 inch cord) 01 a JSO180-4A Headset Adapter (8 foot cord).

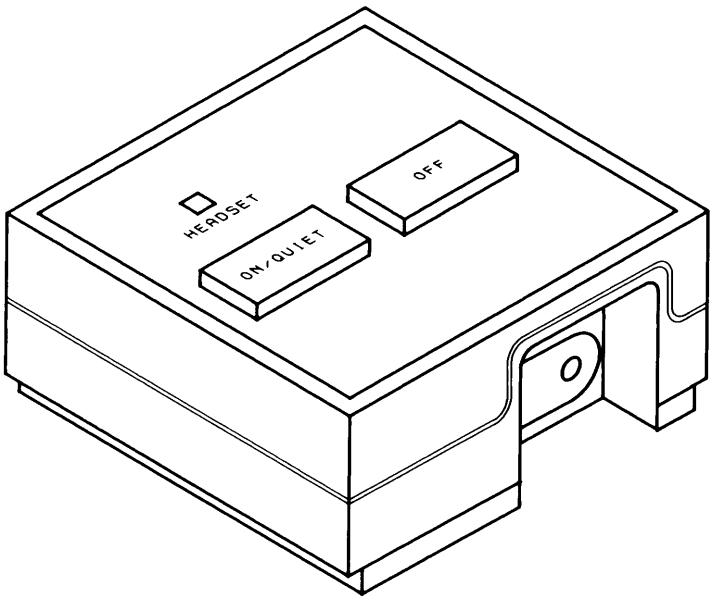
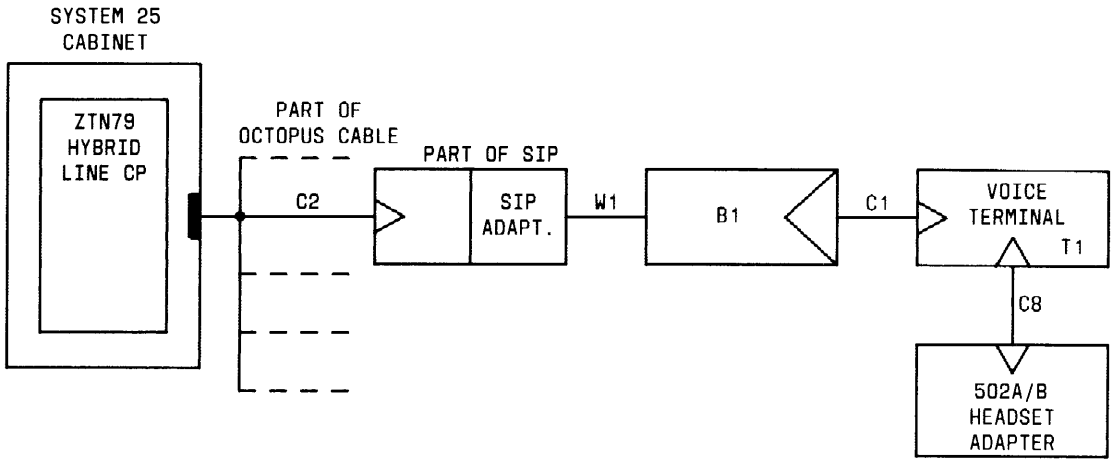


Figure 2-22. 500A/502A/502B Headset Adapter

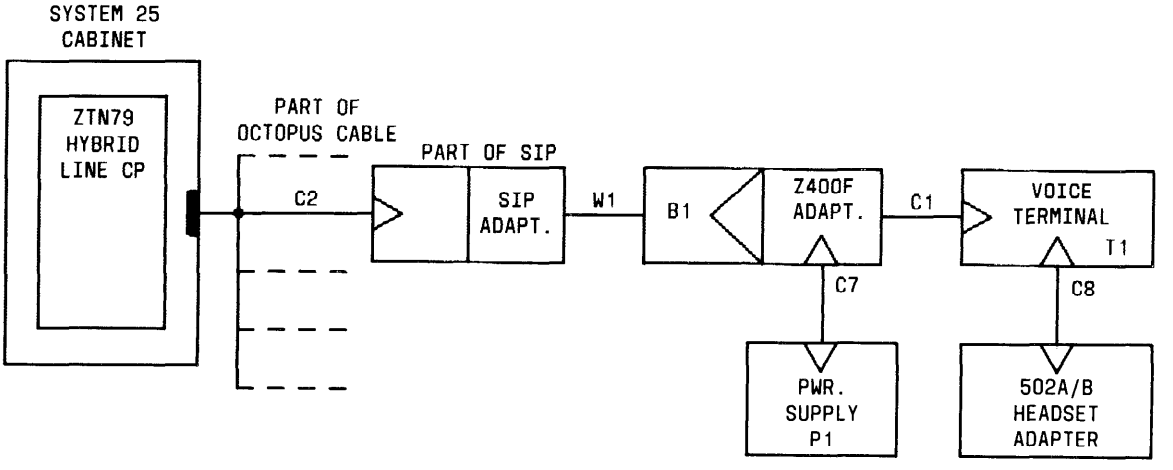


LEGEND :

B1 – TYPICAL-103A CONNECTING BLOCK*
C1 -MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 -OCTOPUS CABLE (WP90780) - PEC 2720-05P
C8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
T1 –7300H SERIES VOICE TERMINAL (EXCEPT 34-BUTTON DELUXE AND ALL SETS WITH BUILT-IN SPEAKERPHONE)
W1 –4-PAIR INSIDE WIRING CABLE*

* - FURNISHED BY INSTALLER

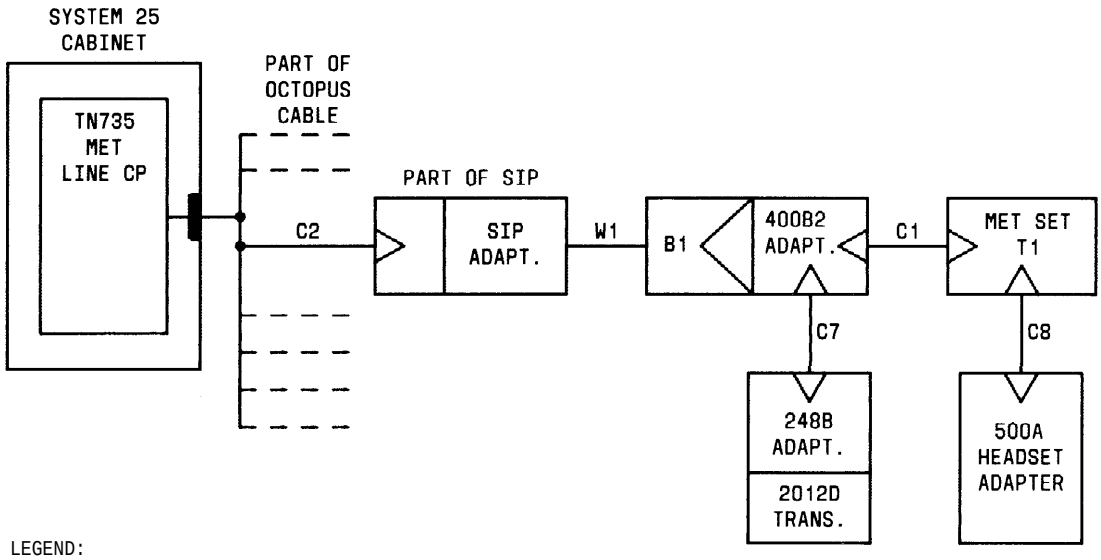
Figure 2-23. Typical Headset Adapter to 7300H Series Voice Terminal Connections Not Requiring Auxiliary Power



LEGEND:

E1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
T1 - 7300H SERIES VOICE TERMINAL (34-BUTTON DELUXE AND ALL SETS WITH BUILT-IN SPEAKERPHONE)
W1 - 4-PAIR INSIDE WIRING CABLE*
C7 - MODULAR CORD (D6AP-87) } PEC 62510
P1 - KS-22911 POWER SUPPLY } PEC 62510
Z400F - ADAPTER
* - FURNISHED BY INSTALLER

Figure 2-24. Typical Headset Adapter to 7300H Series Voice Terminal Connections Requiring Auxiliary Power



LEGEND:

B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) -PEC 2720-05P
C8 - SPECIAL CORD- FURNISHED WITH ADJUNCT
T1 - 7203M -12-BUTTON MET SET
W1 - 4-PAIR INSIDE WIRING CABLE*

248B ADAPTER - MODULARIZES 2012D SOURCE
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE
C7 - MODULAR CORD (D6AP-87)

PEC 21691

* - FURNISHED BY INSTALLER

Figure 2-25. Typical Headset Adapter Connections For 12-Button MET Sets

Hold

Description

This feature allows users to temporarily disconnect from one call and either place or answer another call. A single-line voice terminal user can place only one call on hold and must remain off-hook to retain the held call. A multiline voice terminal user can place as many calls on hold as it has lines and can hang up without losing held calls.

Single-line users can place a call on hold by flashing the switchhook (the user receives Confirmation Tone). The user can then dial another party or return to the held call by flashing the switchhook twice. The first switchhook flash sets up a conference call, the second flash drops the third party; if System 25 Dial Tone, Busy Tone, or Reorder Tone (but not Ringback Tone) was obtained when the third party was dialed, a *single* switchhook flash will drop the tone and return the user to the held party.

Multiline voice terminal users can press HOLD and subsequently replace the handset or call another party without losing the held call. The status LED associated with the held call winks on all terminals with an appearance of the call, except in the case of a conference call. In this case, the wink indication is given only to the party who invoked hold. To return to the held call the multiline user goes off-hook, then presses the call appearance button associated with the held call.

Considerations

The Hold feature allows voice terminal users to handle several calls simultaneously. For single-line sets, placing a call on hold is the first step in transferring or conferencing the call.

Interactions

The following features interact with Hold.

Attendant Console: The Attendant does not receive hold indications for lines (trunks) appearing on the Console unless he/she placed the call on hold.

Bridging of System Access Buttons: A principal or bridging station user who is active on a bridged call can hold the call by pressing the HOLD button. If there is still a bridging or principal station active on the call, the green status LEDs of all associated System Access (SA) and Bridged Access (BA) buttons remain lighted steadily. If no other principal or bridging station is active on the call, the green status LEDs of all associated SA and BA buttons wink.

Any of the principal or bridging stations can enter the held call, unless Exclusion has been activated or the maximum number of parties are already connected to the conversation.

Exclusion: A call can be placed on hold after Exclusion is invoked. The status LED of the line appearance button and the Exclusion button will wink.

Music-On-Hold: A held party on an outside line will receive Music-On-Hold if provided.

Personal Lines: A Personal Line cannot be placed on hold if any other stations are also off-hook on that line.

Remote Access: Remote Access callers cannot use the Hold feature.

Inspection

Description

This feature enables the user at any voice terminal administered for the Display feature to perform the following functions:

- View incoming call identification messages even while busy on another call.
- View information about held or bridged calls.
- Examine information stored on REP DIAL, FLEX DSS, and LAST # DIALED buttons.
- Determine the busy status of pooled trunk groups.
- Determine feature button type.

To activate this feature, the display set user must first leave Normal Mode and enter the Inspection Mode by pressing the INSPECT button. The system presents the following prompt on the display screen:

Screen 1 | INSPECT |

Then, to inspect specific information, the user presses the appropriate call appearance, loop, or feature button.

Note: If the user presses INSPECT immediately after dialing a call, the call may be disconnected; a pause of several seconds is recommended.

Display Formats

When the user presses the button to be inspected, the INSPECT prompt is replaced by information about the button.

- Assigned Feature Button

  The name of the feature is displayed (for example, ACCT ENTRY, SIGNAL, DIRECTORY).

- Unassigned Feature Button

  NOT ASSIGNED is displayed.

- Idle Call Appearance Button

  The type of call appearance is displayed (for example, SYS ACCESS, PERS LINE).

- Active Call Appearance Button

  The normal call information pertaining to the call is displayed. If the display user is excluded, EXCLUDED is displayed.

- FLEX DSS Button

  The stored number and the associated name are displayed. If the button is not programmed, FLEX DSS is displayed.

- REP DIAL Button

  The digits stored on the button are displayed. If the button is not programmed, REP DIAL is displayed.

- FACILITY Button

  The display presents information about the pooled facility in the following form.

Screen 1	XXX OF YYY BUSY

The user can return from Inspection Mode to Normal Mode or Local Mode by any of the following actions:

- Press INSPECT again.
- Allow timeout to occur after 15 seconds of no station activity.
- Change switchhook state; if the user goes on-hook, the terminal returns to Normal or Local Mode; if the user goes off-hook, the terminal returns to Normal Mode.

Considerations

Activation of the Inspection feature has the following impact on terminal operation:

- Hands-Free Answer on Intercom is disabled.
- If the user is on-hook but has a call on hold, there is no effect on the call.
- If the user is off-hook and in the midst of dialing, the system disconnects the call.
- Automatic Incoming Call Identification (see “Display” feature description) is suppressed.
- The user cannot operate feature buttons.
- The user cannot perform call-handling procedures (such as holding, transferring, or answering a call).

The user of a non-attendant display set can exit from Inspection directly to Program Mode by activating the Program switch. However, to reenter Inspection, the user must first go from Program-to Normal, then enter Inspection Mode.

Administration Requirements

Administration of the Display feature enables Inspection.

Hardware Requirements

The Inspection feature can be used only at display-equipped mutiline voice terminals.

Integrated Solution

Description

This feature is the enhancement of System 25 with a UNIX PC acting as a Master Controller. The UNIX PC is a multi-tasking computer which provides the following options:

- Basic Administration (vt100 terminal emulation is a standard feature of the UNIX PC System Software)
- Advanced Administration Software (UNIX) available as a user-friendly alternative to Basic Administration
- Call Accounting System
- Office Automation Available packages include word processing, spreadsheet analysis, and database management
- VOICE POWER Voice Message System. VMS functions include:
  - — Automated Attendant
  - — Voice Mail System
  - — Call Coverage
  - — Announcement Service
  - — Message Drop Service

The UNIX PC/Master Controller may provide any combination of these features simultaneously.

Considerations

For Basic or Advanced Administration, the Master Controller is connected to System 25 exactly as an ordinary SAT. A serial port is required on the UNIX PC (one is standard, additional serial ports are available on expansion boards). Advanced Administration is optional and requires the appropriate software.

For Call Accounting System, the Master Controller is connected to System 25 just as an SMDR printer. The final connection to the UNIX PC is made with a 355A adapter. A serial port is required on the UNIX PC (one is standard, additional serial ports are available on expansion boards). A wide-carriage parallel printer such as the 471 or 479 is strongly recommended (a single Centronics 36 pin parallel port is standard on the UNIX PC).

For Office Automation applications, there are no absolute requirements, but the customer's needs should be understood. For example, most customers who need word processing also would like a letter-quality printer. A serial interfaced printer should be recommended due to the ease with which serial ports may be added to the UNIX PC. Most serial printers will

require a null modem cable such as our PEC 2724-91 G or 2724-92G.

Any VMS application will require at least one VOICE POWER expansion board. Message Drop Service requires a dedicated VOICE POWER expansion board. Announcement Service requires a dedicated VOICE POWER expansion board. Automated Attendant, Voice Mail System, and Call Coverage may share the same expansion board but traffic considerations must determine the number of boards required to provide an acceptable level of service.

The UNIX PC provides three slots for expansion boards. These slots may be used to expand system memory, the number of serial ports, or to add VOICE POWER expansion boards. If more than three boards will be required (a frequent occurrence) an Expansion Unit is available. The Expansion Unit permits a total of seven expansion boards to be used with a single UNIX PC.

Interactions

Both Administration and Call Accounting require a serial port on the UNIX PC. If both will be used an expansion board will be necessary to provide more serial ports. In general it is easier to reconfigure administration to use the second serial port than trying to reconfigure Call Accounting System.

The Automated Attendant feature of VMS may interact with the Call Coverage feature of VMS if Call Coverage is invoked when Automated Attendant attempts to transfer an incoming call. If this happens, two Voice Power boards begin talking to each other and the incoming caller is left needlessly confused. Avoid this situation by administering the number of rings for your Automated Attendant to return to a caller to be fewer than the number of rings it takes to activate Call Coverage. Alternatively, if the station in question will always be available to answer a call, administer the Automated Attendant to perform blind transfers. (To obtain a better understanding of the VMS features and their interactions, read the VMS documentation and the Integrated Solution user guide.)

Administration Requirements

System:

- Requires one port assignment on a ZTN78 Tip Ring Line CP per VOICE POWER expansion board.
- VOICE POWER boards which will be used for Automated Attendant should be administered as a single DGC group.
- Appropriate feature port type codes exist for the different VMS functions and should be administered (see the *R2VI Administration Manual* for the correct data values).

Hardware Requirements

Requires appropriate cables and connectors for Administration and/or Call Accounting

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters are provided under the heading "Connectivity" in section 4 of this manual.

Intercept Treatment With Reorder Tones

Description

Reorder tone (fast busy) is provided when a call cannot be completed (for example when an unassigned or toll-restricted number is dialed, a dialing error occurs, a requested trunk group is busy, or an attempt to park a call fails).

Calls to FPDCs that are not signed-in anywhere or to unassigned DID numbers will be routed to the attendant or will receive Reorder Tone, at the System Administrator's option. Any attempt to dial a restricted call (toll or access restricted) will be intercepted and routed to Reorder Tone.

Considerations

Intercept treatment provides a calling party with positive feedback of an error in dialing or use of an incorrect code.

Interactions

The following feature interacts with Intercept Treatment With Reorder Tones.

Account Code Entry, Forced: The user receives reorder tone when an account code is required on a call but is not entered.

Callback Queuing: If automatic Callback Queuing (CBQ) for outside calls is administered, and all trunks are busy, Queuing Tone is returned to the calling party. If automatic CBQ is not administered, the caller hears Reorder Tone when all trunks in a pool are busy, but can queue the call using the appropriate manual method. Reorder Tone is also returned if the busy trunks are not administered for queuing or if all the queue slots are in use. In these cases, queuing can not take place.

Park: An unsuccessful attempt to park a call due to misdialing or attempting to park more than one call at a voice terminal results in Reorder Tone.

Interdigit Timeouts

Description

This feature allows an originating register to be made available for others if dialing is not completed within a set time period.

Interdigit timeouts is 24 seconds until the first five digits have been dialed, 10 seconds until the next five digits have been dialed, and five seconds thereafter.

After timeout, voice terminal dial pad button presses are interpreted as end-to-end signaling requests and touch-tones are placed on the outside line.

Considerations

Interdigit timeouts also apply to data calls.

When a user dials out over a trunk set up for dial pulse rather that Touch-Tone service, the interdigit timeout interval is involved. The caller cannot speak to the called party until the timeout expires (even though the caller may be able to hear the called party). If the user presses the # button after dialing the last digit this timeout ends and the caller may speak immediately.

Interactions

The following feature interacts with Interdigit Timeouts.

Tie Trunks: For tie trunks only, the talk/signaling path is cut through when answer supervision is received from the distant end. Thus, there is no need to dial # or wait additional seconds for timeout.

Last Number Dialed

Description

Last Number Dialed automatically saves the last number dialed from a multiline voice terminal and allows the user to place the call again without manually redialing the number. The feature is administered to a button on the terminal. Both inside and outside calls can be made in this way. The original call can be placed by manual dialing, by operation of a programmed button, or by Speed Dialing.

To use the feature, the caller first gets dial tone, and then presses the Last Number Dialed button (LAST # DIALED). The l-use and status LEDs of the button selected for originating the call (for example, System Access, Loop, or Personal Line) light steadily; if the Last Number Dialed button has an LED, it lights momentarily. The call proceeds in the normal way. The number associated with the Last Number Dialed button remains saved even if the called party answers. Only the dialing of a new number changes the state of the Last Number Dialed button; the old number is then erased and the new one stored.

Considerations

Last Number Dialed is a convenience feature that is especially valuable for redialing multi-digit numbers that were first dialed manually from a terminal's pushbutton dial. However, the feature can also be used to redial numbers originally called by the following means:

- Repertory Dial buttons.
- Group Select and DXS buttons on a Selector Console.

This feature saves numbers with up to 16 digits.

The user must hear dial tone before pressing the Last Number Dialed button.

The Last Number Dialed feature cannot be activated by dialing an access code. It is not available to users of single-line voice terminals.

If a dialed number does not complete a call, Last Number Dialed still stores the digits dialed. If the user presses the Last Number Dialed button, then dials additional digits to complete the call, both the currently stored digits and the dialed digits will be stored.

Interactions

The following features interact with Last Number Dialed.

Account Code Entry, Forced or Optional: Last Number Dialed does not save the access code *0 or the account code.

Attendant Display: When a call is placed at a Switched Loop Attendant Console, using the Last Number Dialed button, the call information display has the normal format of an outgoing call. In the Inspect mode, pressing the Last Number Dialed button displays the number

currently stored on the button.

Automatic Intercom: Numbers called using an Automatic Intercom button are not saved by the Last Number Dialed feature. The number currently stored by Last Number Dialed is not changed by operations of the Automatic Intercom button.

Bridging of System Access Buttons: The Last Number Dialed feature saves numbers called from either Bridged Access buttons or System Access buttons.

Call Accountability: When a station user dials ##PDC to provide accountability for a call and then dials the desired digits, the ##PDC is not saved by the Last Number Dialed feature.

Conference: When a station user adds a party to a conference, the number dialed is saved as the Last Number Dialed.

Direct Station Selection (DSS): Numbers called using a DSS or Flex DSS button are not saved by the Last Number Dialed feature. The number currently stored by Last Number Dialed is not changed by operations of these buttons.

Personal Lines: When a user originates a call from a Personal Line, only the digits dialed after the line is accessed are saved by the Last Number Dialed feature. The same type of line must be selected to get dial tone for placing another call using Last Number Dialed. If a different type of line is used, the call may be directed to the wrong destination.

Pooled Facility—Direct Access: When a user originates a call from a Direct Facility Access button, only the digits dialed after the line is accessed are saved by the Last Number Dialed feature. The same type of button should be selected to get dial tone for placing another call using Last Number Dialed. If a different type of button is used, the call may not be completed properly.

Repertory Dialing: When using a Repertory Dialing button to place a call, the numbers dialed are saved by Last Number Dialed. When the call is redialed using Last Number Dialed, the same type of button where dial tone was originally accessed should be used again to ensure that the call is directed to the correct destination.

Speed Dialing: When using a Personal or System Speed Dialing code to place a call, the code is saved by Last Number Dialed. When the call is redialed using Last Number Dialed, the same type of button where dial tone was originally accessed should be used again to ensure that the call is completed properly.

System Access/System Access Originate Only Buttons: If a user originates a call from one of these buttons, the same type of button should be selected for getting dial tone to place a second call with the Last Number Dialed feature. Using another type of button, such as Personal Line or Direct Facility Access, to get dial tone may prevent the call from completing properly.

Transfer: When a station user Transfers a call, the dialed number (of the party to whom the call is transferred) is saved as the Last Number Dialed.

Administration Requirements

Last Number Dialed is a default feature on all multiline voice terminals. One button is assigned to the feature at each set. The feature can be moved or removed by administration.

Leave Word Calling

Description

Leave Word Calling (LWC) is available only when a Voice Message System (VMS) is connected to the System 25; the VMS provides the message processing and voice synthesizing facilities used by LWC. The interface between System 25 and VMS requires administration of special ports on analog tip/ring circuit packs.

LWC enhances the messaging capabilities of System 25 by enabling users to generate “call me“ voice messages for PDCs and FPDCs. The messages have a format such as “Call (name) on extension (number)” and are assembled, stored, and delivered by VMS. Called parties are alerted to their messages by lighted Message LEDs, when available.

At a multiline voice terminal, LWC can be used when the terminal is off-hook under any of the following conditions:

- Receiving busy tone
- Receiving ringback tone
- Queued on the called station
- Connected to a coverage point for the called station.

LWC is then activated by pressing the LEAVE WORD button, which sends a request to VMS to leave a message for the called endpoint (station or FPDC).

An LWC message can also be sent when the multiline terminal is not active on a call. In this case, the user goes off-hook to get dial tone, presses LEAVE WORD, and dials the desired extension number.

The single-line voice terminal user who wishes to leave a message at another station must first go from the on-hook state to the off-hook state to get dial tone. Then the user dials LWC code #92 followed by the desired extension number.

in all these procedures, confirmation tone is returned to the caller to indicate acceptance of the dialed code. Reorder tone indicates that the process cannot be completed. After leaving a message, the caller can hang up or handle other calls.

System 25 attempts to deliver LWC requests to VMS as they are generated. Requests are queued if the VMS ports are busy. Up to 20 requests can be queued at one time and be waiting for a voice messaging port to become available. While an LWC request is in queue, the Message LED at the called station will not light and the message cannot be retrieved.

A PDC is administered for users to call to retrieve their messages from the VMS. This PDC is the same one assigned to voice messaging ports and used by the system to interface with peripheral messaging equipment. To retrieve a message indicated by a lighted Message LED, the user dials the message station PDC. All stored messages are presented to the user by voice synthesis. The user can use touch-tone signals to have messages repeated and to erase messages. Password protection is provided.

Considerations

System 25's Leave Word Calling feature provides an easy method to send "call me" 'messages by way of an attached VMS. When the feature is activated, caller and called party information is delivered to the VMS, which lights the Message LED of the called party and then prepares a brief message containing the caller's name and extension. The called party can retrieve the message at his/her convenience.

Multiline voice terminals cannot use dial code #92 to activate LWC.

The VMS system itself must be administered to register the PDCs and FPDCs of all calling and called stations.

Interactions

The following features interact with Leave Word Calling.

Callback Queuing: A user who is queued for access to a busy station can invoke LWC. The callback request is canceled when LWC is activated.

Coverage: A multiline voice terminal user can activate LWC for the called party (PDC or FPDC) even if the call has gone to coverage.

Display: If a user successfully activates LWC, the display shows the called extension number and MSG SENT.

Screen 1	879 MSG SENT

If an LWC attempt is not successful, the display shows the called extension and MSG DENIED.

Screen 1	879 MSG DENIED

Voice Message System: LWC can be used only in conjunction with a VOICE POWER Voice Message System.

Administration Requirements

Multiline Voice Terminal Port:

- Assign a LEAVE WORD feature button.

System:

- Establish interface with associated Voice Message System.

Hardware Requirements

A Voice Message System must be connected to the System 25 by way of a port on a ZTN78 Tip Ring Line circuit pack. Use of a TN742 Analog Line circuit pack for VMS interface is not recommended.

Line Selection

Description

Multiline voice terminals may have many line (facility) appearances. There are three methods by which a user may select a desired line: (1) Prime Line Preference, (2) Ringing Line Preference, and (3) Preselection.

Prime Line Preference:

Automatically connects a multiline voice terminal to a specified line or facility designated as preferred when the terminal goes off-hook. This feature may be assigned to System Access, Loop, Bridged Access, Automatic Intercom, DSS, Personal Line, and Direct Facility Access buttons.

On the Switched Loop Attendant Console, the topmost Loop button has Prime Line Preference by default. However, the feature can be assigned to any of the five Loop buttons.

The user may override this feature by preselecting another button (see below).

If Prime Line Preference is assigned to an Automatic Intercom (AUTO ICOM) or DSS button, the called voice terminal will ring as soon as the terminal goes off hook.

When the Prime Line Preference feature is assigned to an AUTO ICOM, DSS, or Direct Facility Access (FACILITY) button, the button must be pressed to activate the busy-to-idle reminder even though the button's l-Use LED is lighted steadily.

If Prime Line Preference is assigned to a Personal Line or Bridged Access button, the user is connected to the button upon going off-hook, even if the line is busy. The user is always connected to the button where Prime Line Preference is assigned, unless the line has a call on hold.

Ringing Line Preference:

Automatically connects a multiline voice terminal to an incoming call ringing at the terminal.

Ringing Line Preference overrides Prime Line Preference and Preelection when a call is ringing at an on-hook voice terminal.

Line access buttons that can be selected by Ringing Line Preference include System Access, Bridged Access, Automatic Intercom, Coverage, and Personal Lines.

If two or more lines on a multiline terminal or a Direct Trunk Attendant Console are ringing simultaneously, the user is connected to the first line to start ringing. If the user wishes to use a different line, the line *must* be preselected prior to going off-hook. If ringing ceases while the user is on-hook, line preference reverts to whichever option is applicable (Prime Line Preference or no preference).

If a line rings at a multiline terminal when the terminal is busy on another call, Ringing Line Preference will not activate, even if the user goes on-hook during the ringing cycle. However, Ringing Line Preference is not canceled at the Attendant Console while the attendant is off-hook. If a line is ringing while the attendant is off-hook, the ringing line will be selected as soon as the attendant goes on-hook.

Preselection:

Allows multiline voice terminal users to override the above line preference features.

Users may simply press a desired line access button before going off-hook. The user will be connected to the facility selected unless the facility is busy and the party using it has invoked Exclusion or is part of a conference call that is at maximum capacity.

When off-hook, a user can select a facility by pressing the associated button. (This will terminate the call the user was on.)

A user may activate the busy-to-idle reminder on a busy AUTO ICOM, DSS, FLEX DSS, or FACILITY button by pressing the button while on-hook. A burst of ringing is provided when the facility becomes idle. Refer to the description of the Busy-to-Idle feature for additional information.

Considerations

Prime Line Preference (on the topmost SYSTEM ACCESS or Loop button) and Ringing Line Preference are assigned by default to all multiline voice terminals. While these assignments may be changed, it is *strongly recommended* that Ringing Line Preference be retained.

It is recommended that Prime Line Preference not be assigned to a Direct Trunk Attendant Console (DTAC).

Preselection allows users to override line preference features already administered for the terminal and to activate the busy-to-idle reminder feature.

Interactions

The following features interact with Line Selection.

Attendant Console, Direct Trunk: If a line rings at a DTAC while the attendant is on another call, Ringing Line Preference will be invoked when the attendant hangs up.

Bridging of System Access Buttons: A Bridged Access button can be specified as the preferred line for outgoing calls when the station goes off-hook.

If a station has Ringing Line Preference enabled and has a ringing bridged call appearance, an on-hook user is connected to the bridged appearance if the set goes off-hook.

Power Failure/Cold Start: On power-up, most multiline voice terminals will have no l-use LED lit and will not draw dial tone until a button is pressed. After this, line selection should work as described above.

Administration Requirements

Voice Terminal Port:

- Prime Line Preference:
  Assign Prime Line Preference; default = top SYSTEM ACCESS button or top LOOP button (Switched Loop Attendant Console only).

- Ringing Line Preference:
  Assign Ringing Line Preference (yes or no; default = yes).

Line Status and I-Use Indications

Description

Provides users with a visual indication of the status of feature buttons and lines appearing at a their multiline terminals. A green status LED and a red I-Use LED are provided for each programmable button on most-multiline voice terminals.

Table 2-K summarizes LED states and associated descriptions for line appearances.

Table 2-K. LED Indications

I-Use (Red LED)	Line Status (Green LED)	Meaning
Off	Off	Facility is idle.
On	On	If off-hook, facility is in use at this terminal. If on-hook, busy-to-idle reminder is set.
Off	On	Facility is busy or Feature has been activated.
Off	Winking	Facility placed on hold.
On	Flashing	Facility ringing; call will be answered if user goes off-hook.
Off	Flashing	Facility ringing; call will not be answered if user goes off-hook.
On	Off	Facility that will be accessed upon going off-hook.
Off	Broken Flutter	Facility is being transferred or conference.

Considerations

Line Status and I-Use indications provide the user with visible indications of the status of the lines and features.

Interactions

The following features interact with Line Status and I-Use Indications.

Attendant Console, Direct Trunk: When a line that appears at both the attendant position and a multiline voice terminal is placed on hold by the terminal user, the green status LED winks at the terminal but lights steadily on the Attendant Console. When the line is placed on hold by the attendant, the green status LED winks on the console and on voice terminals on which it appears.

Bridging of System Access Buttons: The meanings of green line status and red I-use indications on Bridged Access button are the same as for System Access buttons.

Personal Lines: Trunk-to-trunk transfers will cause the affected PERS LINE buttons on the DTAC to wink.

Local Display

Description

This feature allows the user of a display-equipped voice terminal to operate the Time/Timer circuit built into the display module. The voice terminal must be in the “Local Mode” for use of Local Display. The Timer has **Set, Start, Fwd** (Forward), **Stop, Rev** (Reverse), **Time/Timer,** and **Exit** buttons to control the visible clock, calendar, and 60-minute timer displays and an audible alarm. Some terminals are equipped with a LOCAL button for entering and exiting Local Mode.

Local Display operation depends on the functional assignment and administration of the voice terminal:

- General Use Station or Not-Logged-In Direct Group Calling (DGC) Group Member Station—LOCAL button optional.

  When the terminal is idle, Local Mode is automatically on and the clock/calendar screen is displayed. Any call-handling activity (such as a ringing call or going off-hook) overrides Local Mode and displays the appropriate Normal Mode data. During Normal Mode activity, the user can manually return to Local Mode (for example, to find out what time it is or to time a call) by pressing **Time/Timer;** to return from this condition to Normal Mode, the user presses the active call appearance button. When the call activity ends and the user goes on-hook again, the terminal reverts to Local Mode.

  If the terminal has a LOCAL button, it can be used to turn off Local Mode; during idle periods in this condition, the screen remains completely blank. To return to Local Mode, the LOCAL button must be pressed again.

- Logged-In DGC Group Member Station—LOCAL button optional.

  At a DGC terminal that is logged into the group, Local Mode is normally off. During idle periods, the group queue count is displayed, but the user can press LOCAL, if equipped, or **Time/Timer** to override the queue count and activate Local Display. Any change in queue count, or new call-handling activity, returns the terminal to Normal Mode. When active on a call, the user can go to Local Mode by pressing **Time/Timer** and return to Normal Mode by pressing the active call appearance button.

  If the terminal has a LOCAL button, it can be used exit and enter Local Mode.

- Switched Loop Attendant Console (SLAC)—fixed LOCAL button.

  At a SLAC, Local Mode is normally off. During idle periods, when no calls are being handled and the display screen is blank except for the SLAC queue count, the attendant can press the LOCAL button to override the queue display and enter Local Mode. If the queue count changes or a call comes in, Local Mode is overridden and the queue count or incoming call information (plus queue count) appears on the screen. Initiating a call, or pressing LOCAL again, will also cause the console to exit Local Mode.

The terminal user can set any of the Local Display functions by performing the following procedure while not active on a call:

1. If the clock/calendar screen is not already being displayed, press LOCAL (or **Time/Timer** if the terminal has no LOCAL button).

   The clock/calendar screen appears on the display.

2. Press **Set** repeatedly, until the item to be changed flashes.

3. Press **Fwd** or **Rev** to change the item's setting.

4. To change the setting of another item, return to step 2.

5. Press **Exit.**

   The clock/calendar screen appears on the display.

If the alarm clock function is set, the terminal will "beep" at the selected time. This audible alarm should not be confused with the Attendant System Alarm Indication feature, which causes the ALARM button LED to flash when system trouble is detected.

To time an event (such as a call), the user performs the following procedure while not active on a call:

1. If the clock/calendar screen is not already being displayed, press LOCAL (or **Time/Timer** if the terminal has no LOCAL button).

   The clock/calendar screen appears on the display.

2. Press **Time/Timer.**

   The timer screen appears on the display.

3. To start the timer, press **Start.**

   The timer resets to 00:00, then begins timing.

4. To stop the timer, press **Stop.**

5. To time another event, return to step 3.

6. Press **Exit.**

   The clock/calendar screen appears on the display.

Considerations

At SLACs and logged-in DGC terminals, preference is given to queue displays over the clock/calendar display. This condition can be overridden by pressing LOCAL, if available.

Hardware Requirements

The Local Display feature is available only on display-equipped multiline voice terminals.

Manual Signaling

Description

This feature allows a user to signal another voice terminal. The user may do this at any time, whether on-hook or off-hook. In voice terminal user guides, this feature is called “Signaling.”

Multiline voice terminal users can signal another predesignated *multiline voice* terminal by pressing an associated Manual Signaling (SIGNAL) button. A single tone burst is provided at the signaled terminal. The signaling voice terminal also receives the tone and can use this feature while in any call state. No LED indication is associated with the Manual Signaling feature.

When the Manual Signaling feature is used while the called station is ringing on another call, no audible signal is received by either the signaling or the called voice terminal.

The duration of the single burst of signaling will always be the same, regardless of how long SIGNAL is pressed The signal is repeated each time the button is pressed.

Considerations

Manual Signaling allows a user to signal another voice terminal without calling the terminal. The meaning of the signal may be prearranged between the sending and the receiving parties. Only multiline terminals may be signaled.

Interactions

The following features interact with Manual Signaling.

Display: A display set receiving manual signaling from another station has SIG in positions 1-3. The name of the signaling party, if available, or the extension number of the station from which the signal was sent appears in positions 6-14.

Screen 1	SIG Borden,L

The message is displayed for 5 seconds or until the user selects another button or receives a call. The signaling party has no display for this feature.

External Alerts: Manual Signaling will not activate external alerting devices associated with the signaled station.

Administration Requirements

Voice Terminal Port:

- Assign Manual Signaling (SIGNAL) button.

Message Center-Like Operation (SLAC Only)

Description

A System 25 Switched Loop Attendant Console (SLAC) can be made to function like a message center through administration of call type translations. Certain specific types of calls in the common queue will then be directed only to a console administered as a message center. This arrangement involves no changes in equipment or in operating procedures. The message center attendant answers incoming calls of the preselected types in the normal way. No provision is made for storing messages, a capability often associated with full service message centers.

Message Center Call Types:

The Message Center receives calls of the following types

- Returning parked calls that were originally parked from a Selector Console.
- Returning camped-on calls.
- Returning calls that were extended (transferred) from an Attendant Position to a busy station or a station that does not answer.
- Coverage calls—incoming inside and outside calls (including DID calls) covered by the common queue when the called party does not answer, is busy, or does not want to be disturbed (Send All Calls).
- Floating PDCs (FPDCs) not signed in at a station and unassigned DID calls.

The type of each incoming Message Center call will be identified by a call type descriptor on the console's 16-character display screen. Refer to the earlier Switched Loop Attendant Console feature description for a list of descriptors.

Console Configurations:

Message Center-like operation applies only when the System 25 has two SLACs. In a one-console system, all calls are handled at the same position.

In the default condition, a SLAC is a combined Attendant Position/Message Center. It can receive any type of call. In a one-console system, there is no division between attendant and message center functions.

A dedicated Message Center is a console that is administered to receive only the specified incoming call types. Dial O calls (attendant-seeking calls from inside the system) are not directed to a Message Center, but the console has a unique PDC number that callers can use to reach the attendant.

A dedicated Attendant Position is a console that is administered to answer all of the calls not handled by the Message Center.

Message Center capability can be supported in any of the following two-console configurations:

- One dedicated Attendant Position and one dedicated Message Center. A call extended by the Attendant Position to a station that does not answer or is busy returns to the Message Center.
- One dedicated Attendant Position and one combined Attendant Position/Message Center; the Attendant at the combined position also functions as the Message Center operator for the entire system. A call extended by either Attendant to a station that does not answer or is busy returns to the Message Center.
- Two combined Attendant Position/Message Centers; the Attendants at each combined position also function as Message Center operators. The special call types answered by Message Centers can be divided between the two consoles, or both consoles can answer all types. A call extended by either Attendant to a station that does not answer or in busy returns to the Message Center specified in translation. Three return options are provided: to the 1st attendant, to the 2nd attendant, or to either attendant.

Interactions

All System 25 console features are accessible at a Message Center. Operating procedures are exactly the same as those at a standard Attendant Position. The BIS and HFAI features can be used to answer Message Center calls.

An auxiliary Direct Extension Selector Console can be used with a Message Center. Outgoing calling, from the Basic Console or the Selector Console, is not affected by Message Center administration.

Administration Requirements

A new item in administration allows selection of an alternative set of call type defaults, making one of the attendant positions a “message center”. These defaults may be modified for individual call types, if desired, to tailor the message center-like operation for different locations. For administration purposes, the Message Center call types are divided into four groups: (1) Calls to FPDCs that are not signed in anywhere; (2) Unassigned DID calls; (3) Coverage calls; (4) Returning calls.

With the standard (ie, non-message center) defaults, each of these groups is translated to be directed to all consoles. When both positions are combined Attendant/Message Center consoles, administration can direct specific types to one or both consoles.

Message Center calls are held in the same common queue as any other attendant-seeking calls before being directed to the console.

Messaging Services

Description

These services include features that light a Message LED to indicate that another station (or the attendant) has a message for the user.

The Messaging Services provide light activation/deactivation only. Users must call the sender to receive their messages.

The system supports four types of Message Waiting services:

- Attendant Message Waiting
- Coverage Message Waiting
- Dial Access to Message Waiting Indicators
- Station-to-Station Message Waiting

It also provides interface with the AT&T VOICE POWER Voice Message System.

Attendant Message Waiting

The Attendant can turn on (and turn off) the Message LED at other voice terminals. When this indicator is lighted, users call the attendant for messages. The LED on multiline terminals may be turned off by the user (by pressing MESSAGE), by the attendant, or by another station with the Dial Access to Message Waiting Indicators feature. The Message LED on single-line terminals can be turned on or off by the attendant or by the user with the Dial Access feature.

Refer to the “Attendant Message Waiting” feature description for additional information.

Coverage Message Waiting

Allows a user providing Individual Coverage to control the Message LED on a covered (or bridged) voice terminal. (This feature also allows a bridging station user in a Principal/Bridging arrangement to control the principal’s Message LED using the procedures described here.) A Coverage Message button (COVER MSG) at the covering station is used to display and control the status of the covered user’s Message LED. The state of the COVER MSG LED reflects the state of the covered station’s Message LED. The covering user can turn on or off (toggle) the covered party’s Message LED at any time during a coverage or bridged call by pressing COVER MSG. To turn on the covered user’s Message LED when not on a coverage or bridged call, the covering user may go off-hook on a System Access button, press COVER MSG and then dial the covered user. The covered station’s Message LED turns on if off and stays on if already on. If the covering station then presses COVER MSG a second time before hanging up, the Message LED will turn Off.

A covered party must dial the covering party to retrieve messages. Multiline voice terminal users can press MESSAGE to turn Off their Message LED. Message indicators on single-line voice terminals can be controlled by the user, the covering party, or the attendant, by using the Dial Access feature.

If a user tries to turn on the Message LED at a voice terminal for which the user does not provide Individual Coverage, Reorder Tone is received.

Refer to the “Coverage, Individual” feature description for additional information.

Dial Access to Message Waiting Indicators

This service allows users to turn on or off the built-in Message LED (or Z34A Message Waiting Indicator adjunct) of any voice terminal in the system (including their own). Access is by way of dial codes. The service does not apply to the feature buttons/LEDs administered for Station-to-Station Message Waiting.

Refer to the ''Dial Access to Message Waiting Indicators” feature description for additional information.

Station-To-Station Message Waiting

Multiline voice terminals can be assigned (paired) Message Waiting (MSG WAIT) buttons with associated status LEDs. When this indicator is lighted, the user calls the other user for messages. The MSG WAIT LED can be controlled by the two associated terminals *only;* either user can toggle the state of both LEDs (e.g., both LEDs go On or Off together) at any time, whether on-hook or off-hook.

Refer to the “Station-To-Station Message Waiting” feature description for additional information.

Considerations

The Attendant and Coverage Message Waiting features light the same “basic” Message indicator on each set. The Station-To-Station feature may be assigned to programmable (MSG WAIT) buttons between two sets; it lights the LED next to the button.

Administration Requirements

Attendant Position (Voice Terminal) Port:

- Assign ATT MSG button (defaulted).

Voice Terminal Port:

- Individual Coverage Message Waiting - assign Coverage Message (COVER-MSG) button.
- Assign Individual Coverage (COVER-I ND) between sets.
- Station-To-Station Message Waiting - assign paired Station Message Waiting (MSG WAIT) buttons. Two (multiline) terminals must share this feature.

Hardware Requirements

The Z34A Message Waiting Indicator (MD), if available, can be used on single-line voice terminals not equipped with built-in Message LEDs.

Modem Pooling

Description

Allows switched data connections between digital data endpoints and analog data endpoints.

(Refer also to the discussion of the system's data features provided in the "Data Services Overview" feature description.)

Data transmission between digital and analog endpoints requires a conversion resource since the digital format used by the data module is not compatible with the modulated signals of an analog modem. The conversion resource translates the digital signals from the digital endpoint into analog signals and vice versa.

The modem pool is a single group of up to 12 conversion resources (3 Cabinet system) with the characteristics of a 212A full duplex asynchronous modem that can operate at speeds of 300 and 1200 bps.

The Modem Pooling feature operates transparently to the user whenever possible. In most cases, when a digital endpoint is connected to an analog trunk or port, the system adds a conversion resource without any explicit action by the user.

A voice terminal user who plans to use an analog modem to call a digital endpoint must first enter the Modem Request Code before dialing the digital endpoint. This is because the system assumes that a voice call to a digital endpoint will be transferred to data via the Transfer To Data feature.

A DID call terminating on a digital endpoint will be assigned a modem resource, if available. Otherwise, the call receives Reorder tone.

For each situation that requires a conversion resource, the system:

1. Determines if a resource is required by examining the types of endpoints that are to be connected together or by user indication.

2. Once it is determined that a conversion resource is needed, it is reserved. The user receives Reorder Tone (or the"NO MODEMS - TRY AGAIN" message) if a resource is not available. The system queries the data port to determine whether its options are compatible with those supported by the modem pool. If they are not (e.g., 9600 baud and Permit Mismatch is not enabled), the originating user receives intercept treatment (i.e., INCOMPATIBLE FAR END) and call setup is abandoned.

3. At data connection time, the conversion resource is seized and placed in the connection.

4. The call is disconnected within 15 seconds if the conversion resource does not successfully handshake with both endpoints.

Conversion resources are required for:

- **Data Terminal Dialing:** To establish a data connection for calls originated via Terminal Dialing to intrapremises analog data endpoints.
- **Incoming Trunk Calls:** To establish a data connection between an incoming trunk call and a digital endpoint. Incoming trunk calls that are answered at a voice terminal can be transferred to a data endpoint using the Transfer To Data feature.
- **On-Premises Data Calls:** To establish a data connection between an on-premises analog data endpoint and an on-premises digital endpoint.
- **Outgoing Trunk Calls:** To establish a data connection between an off-premises analog endpoint (modem) and an on-premises digital endpoint.

Considerations

Modem Pooling provides a pool of conversion resources that increases data call flexibility. Conversion resources allow analog data endpoints, using modems, to communicate with digital data endpoints (using data modules). Also, modem pooling reduces costs by sharing resources.

Interactions

The following features interact with Modem Pooling.

Automatic Route Selection: Data calls may be queued. See Table 2-1, “PLEASE WAIT” message.

Calling Restrictions: If a terminal is toll or access restricted, the modem resource is released when the user receives intercept treatment.

Direct Group Calling: Modem pooling supports calls to data endpoints that are part of a DGC group. While an incoming data call is in the DGC group queue, the caller hears Ringback Tone. The conversion resource is inserted if the call is completed to a digital endpoint.

Station Hunting: Modem Pooling supports calls to data endpoints that are part of a station hunting group.

Station Message Detail Recording (SMDR): SMDR records do not reflect modem resource usage. Interpremises data calls using a conversion resource are reported as data calls on the SMDR call record.

Administration Requirements

System (Pooled Modems):

- Modem Request Code (1-9999; default = 820). Allows users to indicate a need for a conversion resource on a data call originated at an analog data endpoint.
- Receiver Responds To Remote Loop (yes or no; default = yes). When active, Data Set Ready is asserted when the modem is in an analog loop test mode.
- Disconnect On Loss Of Carrier (yes or no; default = yes). When active, a loss of the received carrier will cause the modem to terminate the call.
- CF-CB Common (yes or no; default = yes). When active, Clear to Send turns off if Carrier Detect turns off- When a call is being established, Clear to Send and Carrier Detect are not allowed to turn on until carrier has been received and the Clear to Send timer has timed out.
- Disconnect On Received Space (yes or no; default = yes). When active, the modem will disconnect after receiving a “Space” signal of approximately two seconds duration.
- Send Space On Disconnect (yes or no; default = yes). When active, the modem, upon receiving a negation of Data Terminal Ready, sends approximately four seconds of “Space” signal and then disconnects. Without this option active, the modem, upon receiving a negation of the Data Terminal Ready signal, disconnects immediately.

Hardware Requirements

One TN758 Pooled Modem CP provides two conversion resources. Two TN758s are allowed per system cabinet, for a total of 12 conversion resources in a 3-cabinet system.

Music-On-Hold

Description

This feature provides music or other audible indication to a held party on an outside line.

On an outside call, if the user places the call on Hold, or after a call into a DGC group receives the delay announcement, music is provided to the calling party. If a caller receives music because all members of a DGC group are busy, when a group member becomes available to answer the call, music is removed and the calling party is connected to the DGC member.

When a multiline voice terminal user places a call on hold, the status LED of the held line winks and music is provided to the held party. The user may return to the held party by pressing the button associated with the held call. The status LED lights steadily, music is removed from the line, and a talking connection is again established.

When a single-line voice terminal user places a call on hold by pressing the switchhook momentarily, the calling party is connected to music or a recording. The station may return to the held call by pressing the switchhook a second time. The music is removed from the line and the held party is reconnected to the user.

Music-On-Hold is not invoked when a conference call is placed on hold or when the attendant “Start” facility is used to place a call on hold. In addition, Music-On-Hold is not provided for data calls or inside calls.

Considerations

Music-On-Hold lets the waiting party know that he or she is still connected.

Users of equipment that rebroadcasts copyrighted music or other material may be required to obtain a copyright license from a third party such as ASCAP or BMI.

Interactions

The following features interact with Music-On-Hold.

Attendant Splitting One- Way Automatic: Music-On-Hold is not provided when the attendant presses START.

Conference: When a station user sets up a conference, the other parties are put on “special hold” until being connected together. If Music-On-Hold has been administered for the special hold condition, the conferees hear music during the hold interval.

Direct Group Calling: An incoming call to a busy DGC group that provides a recorded delay announcement will receive music after the announcement.

Park: Parked calls (except parked conferences) receive music.

Transfer: When a station user transfers a call to another station, the transferred party is put on "special hold" until the process has been completed. If Music-On-Hold has been administered for the special hold condition, the transferred party hears music during the hold interval.

Administration Requirements

Special Feature Port:

- Assign a port on a TN742 Analog Line or ZTN78 Tip Ring Line CP as required for the music/message source (special feature port type = 254).
- Allow Music-On-Hold for special hold (yes or no).

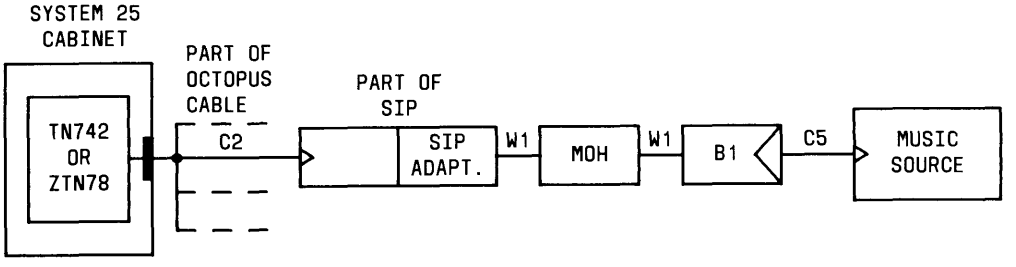
Hardware Requirements

A music source is needed to support the Music-On-Hold feature. The interface is a port on a ZTN78 Tip Ring Line CP or TN742 Analog Line CP.

Detailed connection information is provided in the following figures:

- Figure 2-26— Music-On-Hold Equipment Connections (FCC Registered)
- Figure 2-27— Music-On-Hold Equipment Connections (Non-Registered).

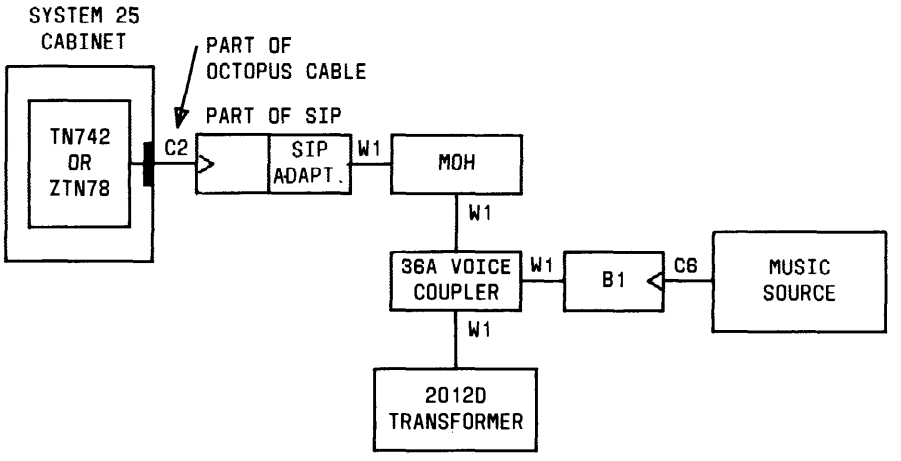
Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading "Connectivity" in Section 4.



LEGEND:

TN742 - ANALOG LINE CP
ZTN78 - TIP RING CP
B1 - TYPICAL-103A CONNECTING BLOCK*
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C5 - MODULAR CORD (D4BU-87)
W1 - 4-PAIR INSIDE WIRING CABLE*
MOH - KS-23395 MUSIC-ON-HOLD INTERFACE UNIT - PEC 62517
* - FURNISHED BY INSTALLER

Figure 2-26. Music-On-Hold Equipment Connections (FCC Registered)



LEGEND:

TN742 - ANALOG LINE CP
ZTN78 - TIP RING CP
B1 - TYPICAL-103A CONNECTING BLOCK*
C2 - OCTOPUS CABLE (WP90780) -PEC 2720-05P
C6 - SINGLE-ENDED MODULAR CORD, D4BY
W1 - 4-PAIR INSIDE WIRING*
MOH - KS-23395 MUSIC-ON-HOLD INTERFACE UNIT
2012D TRANSFORMER - 15-18V AC SOURCE
36A VOICE COUPLER - MUSIC SOURCE INTERFACE
} PEC 62513

* - FURNISHED BY INSTALLER

Figure 2-27. Music-On-Hold Equipment Connections (Non-Registered)

Night Service

Description

Allows users to answer incoming calls on specified trunks when the attendant is not on duty. There are two types of Night Service (NS):

- **Directed NS:** Redirects incoming calls on specified trunks to designated voice terminals.
- **Trunk-Answer-from-Any -Station (TAAS) NS:** Allows users to answer incoming calls on specified trunks by dialing the Night Service access code.

Both types of NS may be provided (specified on a per-trunk basis).

To obtain Night Service, the system must be equipped with an Attendant Console, and the console administered with a NIGHT button. In a system with two Attendant Consoles, both consoles may be assigned a NIGHT button. Either attendant can press NIGHT to activate Night Service. The LEDs of both NIGHT buttons will light to indicate that the system is in the Night Service mode. Pressing NIGHT a second time (by either attendant) deactivates Night Service and turns Off both LEDs.

Directed NS:

Allows an incoming trunk call to be directed to up to four designated voice terminals. Different trunks may be directed to different voice terminals.

When the attendant presses NIGHT, incoming calls on trunks administered to receive Directed NS treatment will automatically be routed to the designated voice terminals (all designated NS stations ring simultaneously). Calls not answered within a specified number of rings will receive a Night Service delay announcement, if available. While at the announcement, they may be bridged onto by going off-hook at a station with a line appearance. The announcement is dropped at this point. If all Directed NS stations for a given trunk are busy (all System Access buttons busy on multiline sets), calls go to the announcement immediately. Directed NS calls do not hunt or receive coverage, but they can be picked up via the Pickup feature.

Personal Line calls that are directed to NS will also ring at the Personal Line appearances and receive normal coverage.

Incoming calls receiving Directed NS treatment will not activate external alerting devices associated with TAAS NS and cannot be answered by dialing the NS access code.

Directed NS is activated under the following conditions:

- An attendant has pressed NIGHT on either console.
- Directed NS has been administered for the trunk.
- Stations have been administered to receive NS calls.

Note that at least one station must be designated as a NS receiver for this feature to work properly. If only an announcement is required, administer the announcement device as a station and make this station the NS receiver.

Refer to the "Night Service Delay Announcement" feature description for additional information on the delay announcement.

Trunk-Answer-from-Any-Station:

Allows any user to answer NS calls. Incoming trunk calls activate an external alerting device such as a bell ("External Alerts" feature). A user can then dial the NS access code and answer the call. Night Service is activated under the following conditions:

- An attendant has pressed NIGHT on either console.
- TAAS NS has been administered for the trunk.
- A NS external alert has been installed and administered.

Note that TAAS NS calls *will not* activate the delay announcement associated with Directed NS.

Considerations

Directed NS provides a means of ensuring that Night Service calls are answered by designating individual voice terminals to receive the calls. In noisy environment, for example, NS via external alerting devices may not be practical. Directed NS provides a solution to the noise problem. Also, Personal Line calls to executives can receive special handling by providing Directed NS. Calls continue to ring at the attendant position or Personal Line appearances when NS is activated. They *also* ring the external alert (TAAS) or Directed NS station.

Trunk-Answer-from-Any-Station provides the capability for any user to answer NS calls.

Interactions

The following features interact with Night Service.

DID Trunks: DID trunks are not assignable to NS. A DID call will ring at the appropriate station whether NS is activated or not.

Following/Forwarding: Directed night service calls will not be given following orforwarding treatment if the PDC is signed in at another station.

Remote Access: Remote Access trunks (dedicated or shared) cannot be given Directed or TAAS Night Service treatment.

Tie Trunks: Dial-in Tie Trunks cannot be given night service treatment.

Administration Requirements

Trunk Ports:

- Assign trunk Class of Service with Night Service (8-15).

  Note: A Remote Access trunk must not be administered for either form of Night Service treatment.

- Assign Directed Night Service trunk (yes or no; default = yes).

Voice Terminal Port:

- Directed NS

  — Add Night Service trunk number to station list.

- Assign External Alert for TAAS NS.

Attendant Console (Voice Terminal) Port:

- Night Service is defaulted to a button on the first Direct Trunk Attendant Console only. On a second Direct Trunk Console or on a Switched Loop Attendant Console, assign Night Service to a flexible button.

- Assign Night Service Access Code.

Hardware Requirements

TAAS NS requires an associated external alert (such as a bell). Each alert requires a port on a ZTN78 Tip Ring Line or a TN742 Analog Line CP. Refer to the “External Alerts” feature description for detailed information and a connection diagram.

Night Service Delay Announcements

Description

This feature provides a recorded announcement for incoming trunk calls when the system has Directed Night Service (NS) activated and a call is not answered.

Directed NS calls not answered within a specified number of rings (1 -15) may be directed to a recorded announcement. Two different recorded announcements may be assigned.

Note that NS calls to a terminal that are not answered do not receive Station Hunting or Coverage treatment (unless the trunk also appears on a station's Personal Line button). After the announcement is played, the call is disconnected.

Considerations

Night Service Delay Announcements provide the calling party with a message that acknowledges the call and can provide additional information as well. Once an NS call goes to the delay announcement, the call will be disconnected from the system after the announcement has been played.

Interactions

The following feature interacts with Night Service Delay Announcements.

Night Service: Incoming calls receiving TAAS NS treatment will not activate the delay announcements. Only trunks that receive Directed NS will activate these announcements.

Administration Requirements

Special Feature Ports:

- Assign first Night Service delay announcement (code 251).
- Assign second Night Service delay announcement (code 252).
- Assign number of rings before Night Service delay announcement (0-15).

Station Ports:

- Assign port circuits (ZTN78 or TN742) for each recorded announcement.

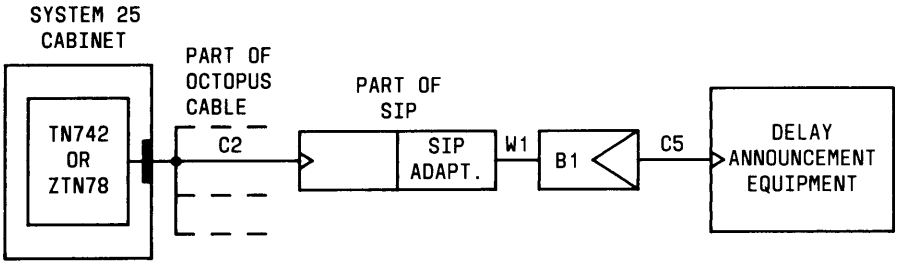
Hardware Requirements

A suitable announcement machine is required for each announcement and must be connected to a port on a ZTN78 Tip Ring Line (or TN742 Analog Line) CP.

The system supports two Directed Night Service delay announcements.

Detailed connection information is provided in Figure 2-28.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading "Connectivity" in Section 4.



LEGEND:

TN742 - ANALOG LINE CP
ZTN78 - TIP RING CP
B1 - TYPICAL-103A CONNECTING BLOCK*
C2 - OCTOPUS CABLE (WP90780) – PEC 2720-05P
C5 - MODULAR CORD (D4BU-87)
W1 - 4-PAIR INSIDE WIRING CABLE*
* - FURNISHED BY INSTALLER

Figure 2-28. Delay Announcement Equipment Connections (FCC Registered)

Off-Premises Stations (OPS)

Description

An Off-Premises Station (OPS) is a single-line voice terminal that is located in another building and connected to System 25 via arrangements with the local CO. The station has the same features as an on-premises single-line station except that it is counted as an outside party in a conference call. Also, the Message feature will not operate with these sets.

Considerations

This service is sometimes furnished to executives at their residences. It allows them remote access to System 25 features and services.

Interactions

The following features interact with Off-Premises Stations.

Conference: For conference purposes, an OPS counts as one of the two allowable outside lines.

Distinctive Ringing: Distinctive Ringing is not provided; OPS stations will always receive standard (that is, single) ringing for calls.

Administration Requirements

Single-Line Voice Terminal Port:

- Assign port on Analog Line (TN742) CP.
- Make This an Extended Station (yes or no; default = np). (This is how the system knows the station is an OPS.)

Hardware Requirements

Requires a port interface on a TN742 Analog Line CP.

The OPS must be a FCC registered single-line voice terminal.

Connectivity information is provided in Section 4, “Hardware Description.”

Out-of-Building Stations

Description

Single-line voice terminals and multiline 7300H series terminals may be directly connected to the system even though they are not located in the same building. For 7300H series terminals, special In-Range Out-of-Building (IROB) units are used to protect the switch and its users from lightning, power crosses, etc. Out-of-Building Stations can access all system features.

Considerations

Single-line voice terminals may be located at distances up to 24000 feet from the system cabinets. Carbon protection devices are required for lightning and power cross protection.

Multiline voice terminals must be located within 2000 feet of the system cabinets (local power required beyond 1000 feet) and must have I ROB protection devices. MET sets may not be used for Out-of-Building service.

Hardware Requirements

Out-of-Building multiline voice terminals require IROB units. Single-line voice terminals require carbon protection devices and must be connected to ports on the TN742 Analog Line CP. Connectivity information is provided in Section 4, “Hardware Description.”

Paging System Access

Description

This feature provides users with dial access or feature button access to paging equipment.

System 25 can provide an interface to paging systems that require a ground start trunk port, a loop start trunk port, or an industry-standard tip/ring station line port. The exact interface depends upon the paging system's requirements. System 25 can also support single-zone, simple amplifier type paging systems that require contact closure (the equivalent of a push-to-talk button) by using a paging system adapter, or by using a special Auxiliary Trunk CP (TN763) with its supporting hardware.

A single-line or multiline voice terminal user (including the attendant) can access paging equipment by dialing the appropriate paging system access code. Depending upon the capabilities and options of the paging system it may be necessary to dial individual zones once the paging equipment has been accessed. (A zone is a number of paging loudspeakers that form a logical group. A zone might include all of the loudspeakers in a given room, or all of the loudspeakers in all of the areas of a building that share a common function. For example, all of the loudspeakers on a factory floor might be grouped into one zone, while all of the loudspeakers in all of the conference rooms might be grouped into another zone.)

In most modern paging systems, paging zones are provided by the paging equipment and selected by passing dial codes to the paging equipment. If you want to retain one or more older paging systems, a similar form of zone paging may be achieved by using an Auxiliary Trunk CP and up to three single zone paging systems. The Auxiliary Trunk CP provides three ports which may be used to interface paging systems. Each of these ports is identified by a unique access code. A fourth code is used to access all three paging systems (zones) simultaneously.

A paging system that is interfaced to System 25 via a ground start trunk port, a loop start trunk port, or an auxiliary trunk port can be administered to be dial restricted. This restricts users from accessing the equipment unless they are assigned DSS buttons with paging access codes. A paging system that is interfaced to System 25 via an industry-standard tip/ring station line port may not be administered to be dial restricted.

Considerations

System 25 can support virtually any new or in-place third-party paging system. Most modern paging systems interface through a ground start trunk port, a loop start trunk port, or an industry-standard tip/ring station line port. System 25 can also support paging systems that require a generic interface with contact closure by using a paging system adapter, or by using an Auxiliary Trunk CP with its supporting hardware. Selection of a particular port as an interface for a paging system will be governed by the requirements of the specific paging system involved.

System 25 supports all AT&T PagePac* voice paging systems. The following are the most common interface arrangements for System 25 and AT&T voice paging systems.

PagePac 20 (with ZoneMate): This paging system interfaces with either a ground start or loop start trunk port. No additional equipment is required. Zone paging is provided by the ZoneMate.

PagePac 20 (without ZoneMate): The PagePac 20 PowerMate may be used alone as a single zone paging amplifier. If the PowerMate is used without a ZoneMate, each PowerMate requires either a port on an Auxiliary Trunk CP (with an Auxiliary Trunk Interface and a Paging/Dial Dictation interface) or a Paging System Adapter. Zone paging is provided by the Auxiliary Trunk CP, if used. Each of the three ports available for paging system access may connect to a paging system, providing up to three paging zones.

PagePac VS: PagePac VS may be interfaced with System 25 by means of a loop start trunk port or a tip/ring station line port. PagePac VS provides a built-in interface card which cannot be removed or changed. No other interface options are supported for PagePac VS.

PagePac Voice Paging Systems: The larger, complete paging systems in our product line are collectively known as PagePac Voice Paging Systems. They are currently available in sizes ranging from 50 Watts to 200 Watts and, when ordered as complete systems, are shipped with a Type C Applique. This Type C Applique may be optioned during installation to interface with a ground start trunk port, loop start trunk port, or a tip/ring station line port.

PagePac Voice Paging Amplicenter: The Amplicenter is the amplifier component of the complete PagePac Voice Paging System. If the Amplicenter is used alone as a single zone paging amplifier it requires either a port on an Auxiliary Trunk CP (with an Auxiliary Trunk Interface and a Paging/Dial Dictation Interface) or a Paging System Adapter. Zone paging is provided by the Auxiliary Trunk CP, if used with multiple Amplicenter. Each of the three ports available for paging system access may connect to a paging system, providing up to three paging zones.

Paging is particularly useful when used in conjunction with the Park feature. When a user is away from his or her location and receives a call, the call can be answered and parked by another user. The called party can then be paged and told what extension number to call to retrieve the parked call. The called party can then retrieve the call from any voice terminal.

Interactions

The following features interact with Paging System Access.

Direct Inward Dialing (DID): A DID call may access a paging code. This allows an outside user to dial in and utilize the Paging System. Dial restricting the paging code will block this interaction.

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* Trademark of Harris Corporation Dracon Division

Tie Trunks: Incoming Tie Trunk calls can access paging ports connected to Auxiliary Trunk circuit packs. If administered as a Tandem Trunk, an incoming Tie Trunk can also access other types of paging ports.

Administration Requirements

Ground Start or Loop Start Trunk Port:

- Requires a port on a ZTN76 Ground Start Trunk CP or ZTN77 Loop Start Trunk CP for each interface required. If the paging equipment requires a contact closure (equivalent to a push-to-talk switch) a paging system adapter must be installed between the port and the paging system.
- Assign as Paging Port. (Action = 16; 1 = Yes or 0 = No). Default = No.
- Assign Trunk Access Code.
- Dial restrict zone (yes or no).
- Assign other appropriate CO trunk parameters.

Auxiliary Trunk Port:

- A port on a TN763 Auxiliary Trunk CP, an Auxiliary Trunk Interface and a Paging/Dial Dictation Interface are required for each interface.
- Assign Paging access code for each paging system (maximum = 3) to be provided.
- Dial restrict each paging system (yes or no)
- Assign an "All Zones" access code.

Multiline Voice Terminal Port:

- Assign DSS button with paging zone access code as required.

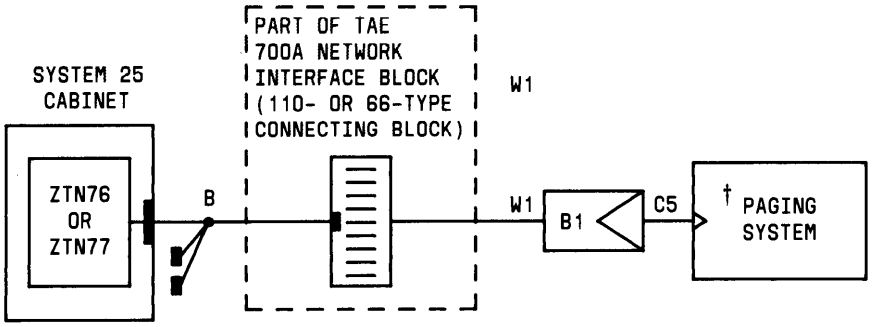
Hardware Requirements

Requires a paging system that is compatible with telephone systems. Each paging system requires at least one suitable System 25 port. Selection of the type of System 25 port (loop start, ground start, Auxiliary trunk or tip/ring station line port) is dependent on the paging system or its adapter. If a choice is available, the recommended method of interfacing is via CO trunk ports (either loop or ground start).

Some customer-provided equipment, typically older systems, may require contact closure to control the paging equipment. in this case either a paging adapter or a port on a TN763 Auxiliary Trunk CP must be used. If an Auxiliary Trunk CP is used to support multiple paging systems (up to three) each paging system constitutes a single zone and an additional dial code may be administered to access all zones simultaneously.

Detailed connection information is provided in Figures 2-29 and 2-30.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters are provided under the heading “Connectivity" in section 4 of this manual.



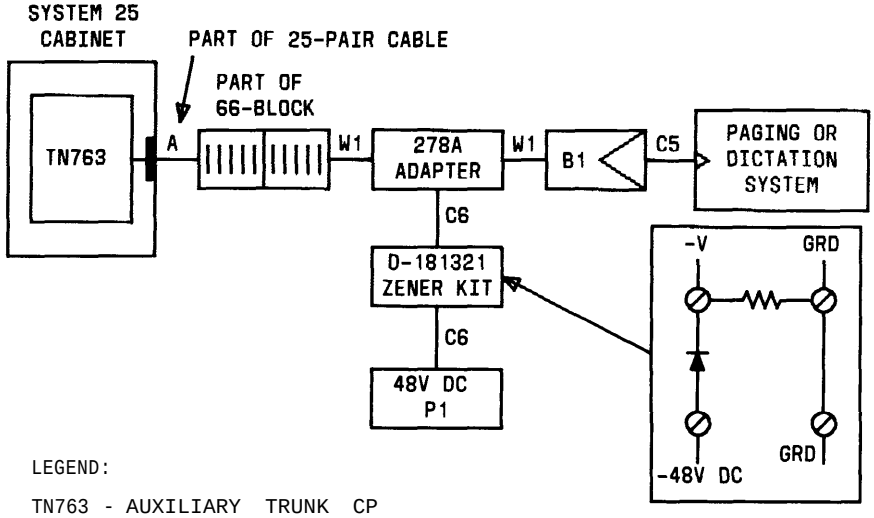
LEGEND:

ZTN76 - CO GROUND START TRUNK CP
ZTN77 - CO LOOP START TRUNK CP
B - 3 TO 1 SPLITTER CABLE-CONNECTORIZED (OR6016) – PEC 2720-06X
B1 - TYPICAL-103A CONNECTING BLOCK*
C5 - MODULAR CORD (D4BU-87)
W1 - 4-PAIR INSIDE WIRING CABLE*

* FURNISHED BY INSTALLER

† PAGING SYSTEM - PAGE PAC 20 E/W ZONE MATE 9 - PROVIDES 9 PAGING ZONES, PLUS ALL-ZONE PAGING

Figure 2-29. Paging Equipment Connections Using CO Trunk Ports (FCC Registered)



LEGEND:

TN763 - AUXILIARY TRUNK CP
A - SINGLE-ENDED 25-PAIR CONNECTOR CABLE (A25D) (NOTE 1)
B1 - TYPICAL - 103A CONNECTING BLOCK*
C5 - MODULAR CORD (D4BU-87)*
C6 - SINGLE-ENDED MODULAR CORD (DYB4) (NOTE 2)
P1 - KS-22911, L1, POWER SUPPLY, 48 VOLT DC (NOTE 2)
W1 - 4-PAIR INSIDE WIRING CABLE*

NOTES:
1. APPARATUS CODE D-181523 (PEC 62511) INCLUDES 66E3-25 BLOCK CONNECTOR AND CABLE B25A 15/DE.
2. APPARATUS CODE D-181524 (PEG 62512) INCLUDES C6, P1, 278A ADAPTER AND ZENER KIT.

* FURNISHED BY INSTALLER

Figure 2-30. Paging Equipment Connection to TN763 Causing 278A Adapter

Park

Description

This feature allows a user to put a call into a special hold/parked condition so that it can be picked up from any voice terminal in the system. It is used in three typical applications:

- Basic Park: A user parks a call and then picks it up at another voice terminal.
- Meet-Me-Conference: A conference member parks the conference and pages another employee to join the conference.
- Transfer: A user parks a call and then pages another employee to pick up the call.

A user parks a call by first putting it on hold and then dialing the Park code (*5). The call can subsequently be retrieved from any voice terminal by dialing the Park retrieval code (*8) and the PDC of the parking station. In addition, any user active in a conference involving fewer than five members may park the conference so that another user may join by dialing *8 and the parked-on number.

A multiline voice terminal user invokes Park by pressing HOLD to place a call or conference on hold, then pressing an idle System Access button and dialing *5. A single-line voice terminal user invokes the feature by pressing the switchhook to place the call or conference on hold, then dialing *5. If the call is successfully parked, the user receives confirmation tone and then recall dial tone. If the call cannot be parked, reorder tone is received. In the latter case, to return to the held call, the user presses the held call button (multiline sets) or flashes the switchhook (single-line sets).

The parking station may return to a parked call or conference without affecting the park state. The multiline voice terminal user may return by pressing the held call button. The single-line user may return by flashing the switchhook.

When the single-line user goes on-hook, the parked call is removed from the terminal and cannot be reentered.

To retrieve a parked call, a user must obtain system dial tone, dial *8 and then dial the PDC of the station that parked the call. If the call is not retrieved within an administered interval (default = 2 minutes) the call will return to the user that parked the call. At multiline voice terminals, returning calls always ring at System Access (SA) buttons, regardless of the type of button on which the parked call arrived originally. If no idle SA button is available, calls attempting to return will remain parked until one becomes idle.

Note: Multiline voice terminals without SA buttons cannot park calls (they receive reorder tone when they try to do so). However, in a principal station/bridging station arrangement, a bridging station without SA buttons can park calls on its Bridged Access buttons because returning calls would ring at the principal station.

An attendant can park a call with the same procedure as a multiline voice terminal user. In addition, if the Attendant Position is equipped with a Selector Console, up to eight trunk calls can be parked on DXS buttons dedicated to the Park function. The attendant parks a call by

pressing an idle Park button. The status LED of the parked line on the Attendant Console winks and the status LED of the Park button on the Selector Console lights steadily. A call parked with the Selector Console is retrieved by dialing *8 and the access code assigned to the dedicated Park button.

A call parked by the attendant using the same procedure as a multiline voice terminal user will return to an SA button on a Direct Trunk Attendant Console (DTAC) or a LOOP button on a Switched Loop Attendant Console (SLAC) if it is not picked up within the administered return interval. A call parked on the Selector Console but not picked up within the interval will return to the RTN-DA button (DTAC) or a LOOP button (SLAC), in the same manner as any other unanswered call.

Each voice terminal user (except an attendant with a Selector Console) can only park one call at a time; a maximum of 24 calls can be parked in the system at one time. A call is no longer parked when it is answered, when it returns to the parking terminal, or when it is abandoned by the caller.

Considerations

Park can be used whenever a user engaged on a call needs to go elsewhere, and wishes to complete the call from another terminal. Park also allows users to answer a call from any voice terminal when paged.

In order to use the Park feature, a station must have at least one System Access button.

Interactions

The following features interact with Park.

Attendant Direct Extension Selection: Station-to-Station calls cannot be parked via the Park buttons on the Attendant Selector Console.

Attendant Position Busy: If a call is parked on an attendant console and the attendant console enters Position Busy mode, the parked call will return to the inactive console if not answered within the administered interval (default = 2 minutes).

If a call is parked on the Selector Console by a Switched Loop attendant and the SLAC is placed in the Position Busy mode, the parked call will return to the other active attendant console, if not answered within the administered interval.

Bridging of System Access Buttons: A call parked by a principal station having bridged call appearances at a bridging station must be retrieved by dialing *8 and the PDC of the principal station. The principal station and the bridging station can enter the call without affecting its parked state. If the call returns, it can be answered at the principal station or on a bridged appearance.

If a bridging station is active on a bridged call appearance and activates Park, the call is parked on the Personal Dial Code (PDC) number of the *principal* station, not of the bridging station. If the parked call is not answered, it will return on the principal's SA button.

Callback Queuing: Queued calls cannot be parked unless they are part of a conference. Reorder tone is returned whenever an illegal park is attempted, but the queued call is not disconnected. If parked conference members drop out, leaving only a queued call, it will be disconnected to prevent the illegal condition of a single queued call being parked.

Calling Restrictions: If the parking station is outward restricted or toll restricted, the recall dial tone following a successful park cannot be used to avoid the restriction.

Conference: Parked conference calls do not return to the parking voice terminal. They remain parked. Park may be used to place a conference on hold if it contains fewer than five parties.

Display: When a parked call returns to a display station, screen 1 contains redirection symbol "}" and PARK RTN; screen 2 contains calling party identification.

Exclusion: A call cannot be parked, and a parked call cannot be answered, if the Exclusion feature is invoked on that call.

Intercept Treatment With Reorder Tone: An unsuccessful attempt to park a call due to misdialing or attempting to park more than one call at a voice terminal results in Reorder Tone.

Music-On-Hold: Parked calls (except conferences) receive music.

Personal Line: A parked Personal Line is bridgeable by any user with a button appearance of that line. Bridging on to the connection does not answer the parked call. The parked call will not return to the parking voice terminal user in this case.

Remote Access: Remote Access callers cannot use the Park feature.

Transfer: Single-line voice terminals cannot transfer parked calls.

Administration Requirements

Special Feature Port, Attendant Selector Console:

- Assign selector console Park codes.
- Number of seconds before Park return (0-240; default = 120).

Personal Dial Code (PDC)

Description

A Personal Dial Code (PDC) consisting of one to four digits is assigned to each voice terminal in the system. The PDC serves as the "extension number" of its terminal. Each PDC can also be associated, through system administration, with the name of the terminal's user; terminals that have no specific users, such as lobby or conference room sets, can be assigned appropriate place or function names.

A special quality of PDCs is their portability; users can "carry" their PDCs with them when they temporarily go to another terminal. A user can "sign in" his or her PDC at "away" terminals and receive calls originally directed to the home terminal. Upon leaving the temporary location, the user"signs out" the PDC so that calls will again ring at the home terminal. (For information on the applications of these procedures, refer to the "Following" feature description.)

Analog data endpoints with modems are also assigned PDCs. Digital data endpoints have Data Dial Codes (DDCs).

Floating Personal Dial Codes (FPDCs) are assigned to users who do not have their own voice terminals. These users can sign in their FPDCs at any station in the system and receive their calls there. The system can be administered to force calls to FPDCs that are not signed in anywhere to ring at the Attendant Console(s). FPDCs can also be reserved and assigned to visitors who expect to receive calls.

Considerations

The Personal Dial Code (PDC) feature provides flexibility for users and visitors. Visitors, once assigned an FPDC, can inform callers and the attendant. Calls can then be directed to the voice terminal where the FPDC is signed in. Calls to FPDCs that are not signed in anywhere may be directed to the attendant for further handling.

Up to 200 PDCs and 300 FPDCs can be assigned in a system.

Interactions

The following features interact with Personal Dial Code.

Coverage: Calls to an FPDC signed in at a voice terminal receive the coverage of that terminal. Unanswered calls to a PDC at an away terminal return to the home terminal and receive the home terminal's coverage treatment; they do not receive the away terminal's coverage.

Direct Inward Dialing: In systems with DID service, PDCs, FPDCs, DGC access codes, DDCs, and facility access codes may match the last 2, 3 or 4 digits of DID numbers. For example, the code matching DID number 555-2345 may be 45, 345 or 2345, depending on the system dial plan.

Direct Station Selection: If an attempt is made to program an FPDC (rather than a PDC) on a Flex DSS button, Reorder Tone is received.

Display: Calls to an FPDC signed in at a station covered by an attendant console receive coverage. However, the attendant display will show the PDC and name of the covered station, not the FPDC.

Administration Requirements

System:

- Route calls for unassigned DID numbers to the Attendant (yes or no; default = yes).
- Route calls for not-signed-in FPDCs to the attendant (yes or no; default = yes).
- Add/Delete FPDCs.

Personal Lines

Description

This feature provides a dedicated outside line for multiline voice terminal users.

Unlike pooled facilities, which can be accessed via dial codes, Personal Lines can be accessed only via a dedicated feature button, and provide both incoming and outgoing service. Up to 16 terminals may share a Personal Line.Up to four parties may be off-hook on the line at the same time (the line itself is the fifth conferee). When the line is busy, its status LED lights at all terminals on which the line appears. Ringing may be provided optionally to one or more of the terminals sharing the line.

For each Personal Line, one station is administered as the principal (owner). The coverage of that terminal determines the coverage of the Personal Line.

Considerations

Personal Lines provide facilities to users who desire direct access to the exchange network. In addition, Personal Line appearances are provided on the Direct Trunk Attendant Console for general use trunks. Appearances of these lines may also be provided at selected multiline voice terminals to ensure coverage when the attendant is not available. DID trunks cannot be terminated on Personal Line buttons.

Personal Lines provide direct access for callers, bypassing the attendant. In some cases, they may substitute for DID service.

Interactions

The following features interact with Personal Line.

Attendant Console, Direct Trunk: On the Direct Trunk Attendant Console (DTAC), trunks are terminated as Personal Lines. The DTAC can accommodate a maximum of 26 Personal Lines (24 is the practical limit).

Attendant Console, Switched Loop: Personal Lines cannot be terminated on a Switched Loop Attendant Console.

Coverage: The coverage of the principal station (owner) determines coverage for the line.

Direct Group Calling: The same trunk may be used as a Personal Line and also be directed to a DGC group. If an incoming call is not answered by the DGC group after a predetermined number of rings, ringing and LED flashing will be transferred to all button appearances of the line (unless a DGC delay announcement is provided).

Hold: A Personal Line cannot be placed on hold if any other stations that share the line are also off-hook on the line.

Line Selection (Prime Line Preference): Prime Line Preference may be assigned to a Personal Line.

Park: **A** parked Personal Line is bridgeable by any user with a button appearance of that line. Bridging on to the connection does not unpark the call; in this case, the parked call will not return to the parking user.

Pickup: After a call is picked up from a Personal Line button, the called terminal can still enter the call.

Pooled Facility: A Personal Line may also be a member of a pooled facility group.

Toll Restriction (see "Calling Restrictions"): A call over a Personal Line is subject to the toll restrictions of the station on which the call was placed.

Administration Requirements

Voice Terminal Port:

- Assign Personal Line trunk number.
- Assign Personal Line feature button.
- Make this the Principal Station (yes or no).
- Enable Personal Line Ringing (yes or no).

Hardware Requirements

Requires port assignments for each trunk interface to be provided and a button termination on multiline voice terminals.

Pickup

Description

This feature allows a user to answer a call ringing at another voice terminal. There are two forms of Pickup: (1) Directed and (2) Group.

Directed Pickup:

Directed Pickup allows calls to other voice terminals, including Automatic Intercom calls and calls ringing at coverage buttons, to be “picked up” by dialing the Pickup code (*7) and the ringing terminal’s PDC. Picked-up calls remain accessible at the call appearance button of multiline terminals, but are no longer available at single-line terminals.

A ringing call can be answered at a busy single-line voice terminal by pressing the switchhook, which will place the current call on hold, dialing *7, and the ringing voice terminal’s PDC.

This is a standard feature available at every voice terminal. No administration is required. Also, this feature cannot be turned off or restricted.

Group Pickup:

Group Pickup permits calls to another terminal in the Pickup group to be answered. Any ringing call, including Automatic Intercom and coverage calls, is eligible for Pickup.

A member of a Pickup Group can answer any call to any other member of the group by dialing the Group Pickup code (’70).

Up to 16 groups (with up to 16 voice terminals in each group) can be set up. Each Pickup group can have a maximum of two simultaneous ringing calls eligible for Pickup treatment at a time, and the calls are picked up in order of arrival. A user can be assigned to only one Pickup Group.

Considerations

With Pickup, users do not have to leave their own voice terminal to answer a call at a nearby voice terminal. Instead, a user simply lifts the handset and dials an access code. This allows calls that may go unanswered to be handled quickly and efficiently.

The call must be administered to *ring* at the voice terminal for which Pickup is attempted. Otherwise the attempt will be blocked.

If the picked-up call was to a multiline terminal, the called terminal can still enter the call. If the called terminal was a single-line terminal, it cannot enter the call once it is picked up.

Pickup cannot be invoked after the call has been answered.

If no eligible calls are ringing, attempting pickup results in reorder tone.

Interactions

The following features interact with Pickup.

Bridging of System Access Buttons: Pickup is a station-oriented feature. Thus, calls ringing at a principal System Access (SA) button can be picked up by members of the principal's Pickup Group; calls ringing at a Bridged Access (BA) button can be picked up by members of the bridging station's Pickup Group. If a user dials the Group Pickup access code while active on a BA button, the system interprets this as an attempt to pick up a call in the Pickup Group of the bridging station, not of the principal station.

Calls ringing at either a principal SA button or an associated BA button can be picked up by using the Directed Pickup feature.

Callback Queuing: A callback call cannot be picked up.

Coverage: When a call is directed to a coverage station and the call is answered via Pickup, all Cover buttons associated with the call go idle.

Personal Line: After a call is picked up from a Personal Line (PERS LINE) button, the called terminal can still enter the call.

Remote Access: Remote Access callers cannot use the Pickup feature.

Administration Requirements

Voice Terminal Port:

- Assign Pickup Group Number (1-16, none; default = none).

Pooled Facility - Dial Access

Description

This feature allows both multiline and single-line voice terminal users to access a common pool of trunks by dialing a facility access code.

Up to 16 facility access codes can be assigned (one per trunk group). The codes can be one to four digits in length. A group of similar trunks assigned the same access code is referred to as a trunk group. Additional information is provided in the "Trunk Groups" feature description.

After going off-hook on a System Access or Loop button, receiving system dial tone, and dialing a facility access code, the user will be connected to an idle trunk. (However, the connection will not be made if the terminal is restricted from dialing this trunk group or if dial access is restricted, in general, to trunks in the group.) The LEDs associated with the System Access button will be lighted, and the user may complete the call. Single-line users do not receive LED indications of the status of the pool. An attempt to originate a call on a busy facility will result in Reorder Tone (fast busy).

Considerations

Pooled Facility-Dial Access provides users of single-line terminals, or multiline voice terminals without Direct Facility Access (FACILITY) buttons, access to the system's pooled facilities.

Interactions

The following features interact with Pooled Facility-Dial Access.

Bridging of System Access Buttons: A station originating a call on a Bridged Access button and using a facility access code is granted access to that pool according to the Calling Restrictions assigned to the bridging station, not the principal station.

Direct Inward Dialing (DID): Access to pooled facilities via DID is permitted. This includes access to WATS, FX, tie trunks, private lines, dictation equipment, and paging systems. This access is provided by selecting facility access codes so that they will match DID numbers.

Outward and Facility Access Restriction (see "Calling Restrictions"): A terminal can be denied dial access to some or all pooled facilities, or may be totally restricted from making any outside calls.

Toll Restriction (see "Calling Restrictions"): Denies the use of pooled facilities for certain toll calls, but does not block access to the pooled facilities.

Administration Requirements

Voice Terminal Port:

- Restrictions - Refer to “Calling Restrictions” feature description.

Trunk Port:

- Assign facility access codes.
- Allow dial access for facility (yes or no).

Pooled Facility - Direct Access

Description

This feature allows multiline voice terminal users to access a common pool of trunks via a Direct Facility Access (FACILITY) button.

Upon pressing a FACILITY button and going off-hook, a multiline voice terminal user is connected to a common pool of outside trunks (i.e., CO, FX, WATS, tie). If the Status and I-Use LEDs associated with the button light steadily, the user may complete the call. If no idle trunk is available (facility busy indication), an attempt by the user to originate a call will be denied and the l-Use indicator will be Off. A user requiring access to several different trunk pools must have a separate FACILITY button for each pool.

If all trunks in a pool are busy, the Status LED will be lighted. The user may press FACILITY and remain on-hook to receive the busy-to-idle reminder when a trunk becomes available. The busy-to-idle reminder is a short burst of tone that will be heard when a trunk in the pool becomes available. When Prime Line Preference is assigned to a FACILITY button, the button must be pressed to invoke the busy-to-idle reminder, even though the l-Use LED is lighted.

Refer to the “Trunk Groups” feature description for additional information.

Considerations

Pooled Facility-Direct Access provides easy access to the exchange network for users who make many outside calls. The feature eliminates the need to dial a facility access code. In addition, the associated status LED provides pool busy/idle status and the busy-to-idle reminder.

Interactions

The following features interact with Pooled Facility-Direct Access.

Automatic Route Selection (ARS): Multiline voice terminal users who have pressed FACILITY to activate busy-to-idle reminder must wait until all queued ARS users have been serviced.

Facility Access Restriction (see “Calling Restrictions”): A trunk group may be reserved for a group of users by dial-access restricting the trunks. In this way, only users who have a FACILITY button, a Personal Line appearance, or who use ARS can use the trunks.

Line Selection (Prime Line Preference): Pressing a FACILITY button to invoke the busy-to-idle reminder overrides Prime Line Preference.

Toll Restriction (see “Calling Restrictions”): Toll-restricted voice terminals receive standard toll restriction treatment on all FACILITY buttons.

Administration Requirements

Voice Terminal Port:

- Assign Direct Facility Access (FACILITY) buttons.

Trunks:

- Assign Facility Access Codes.

Power Failure Transfer (PFT)

Description

This feature provides service to and from the CO for a limited number of prearranged *single-line* voice terminals during a commercial power failure (or when voltage drops below 90 volts for longer than 250 milliseconds) and during other service interruptions. Any loop start or ground start trunk may be arranged to terminate at a specific station on a one-to-one basis. When a failure occurs, these prearranged connections are made, bypassing the system and connecting terminals directly to the CO trunks. System features and restrictions are not available during this time.

The system supports up to four Emergency Transfer Units (ETUs). Each ETU can provide up to five voice terminals with direct connection to CO trunks.

When the system connects to dial pulse trunks, only rotary sets may be used to support Power Failure Transfer (PFT). When the system interfaces the CO via touch-tone trunks, touch-tone single-line voice terminals are used as PFT stations.

When power is restored, the following will be restored to their previous state:

1. Night Service mode (on or off).
2. User-programmed Flex DSS numbers.
3. PDCs signed in at a “home station” or an “away station” remain signed in there; if Forwarding has been activated for a station, it remains in effect.
4. If a voice terminal has been removed/not removed from a DGC group, the terminal will remain in that state.
5. User-programmed Repertory Dialing numbers.
6. All system/station features programmed through system administration.

Considerations

Power Failure Transfer provides emergency incoming and outgoing telecommunications service to a number of predesignated single-line voice terminals. This is particularly important for organizations providing public services such as fire, police, medical, etc.

Hardware Requirements

The 10B Emergency Transfer Unit (ETU) in Figure 2-31 supports up to five Power Failure Transfer (PFT) sets and a DID make-busy function. Up to four ETUs can be supported for a maximum of 20 PFT sets. The sets can be connected to selected Loop Start or Ground Start trunks. If Ground Start trunks are used, a 55A1 Ground Start Button must be provided at each PFT set.

Only FCC registered single-line voice terminals may be used for PFT stations. Rotary sets must be used for dial pulse PFT trunks; touch-tone sets must be used for touch-tone PFT trunks.

In the event of a Power Failure Transfer (switch has lost power or a major fault has occurred) a contact closure is provided to the Central Office (CO) over a dedicated pair of wires. The CO then makes busy all DID trunks. When power is restored, the closure is removed and the CO restores DID service. External alarm contacts are provided on the front of the ETU for use as required.

Note: It is recommended that customers with DID service make provisions with their CO to provide this arrangement.

The ETUs are mounted on the cross-connect backboard. Connections are via 25-pair receptacle-ended (CO and SIP) and plug-ended (switch line and trunk) connectors. Modular jacks are provided for the -48V control signal from the CPU (Call Processor Unit) and for additional ETUs. Screw terminals are provided for the connection of external alarms.

When calculating Unit Loading (see Section 5, “Unit Loads”), all ETU loading counts against Cabinet 1.

The 10B ETU is mounted on the cross-connect field as shown in Section 6, “Environmental Requirements. ”

ETU Power Failure Transfer connections are shown in Figure 2-32. Part (a) on the figure shows a single-line voice terminal that has been connected as a Power Failure Transfer station. In normal operation, the Call Processor CP supplies -48V dc to the ETU. The voice terminal is connected through the ETU to the station port CP and can support all calling activities. The trunk connection through the ETU to the trunk port supports normal trunk calls.

Part (b) on Figure 2-32 shows the ETU connections when a Power Failure Transfer has occurred. The transfer is initiated by the removal of the -48V dc to the ETU. All connections through the system are dropped, and direct connections between PFT stations and CO trunks are established. A contact closure toward the CO makes all DID trunks busy.

When the system is again able to process calls, normal service is automatically restored.

A multiple ETU arrangement is shown in Figure 2-33. As discussed earlier, separate -48V dc control signals from the Call Processor are provided via legs 7 and 8 on Octopus cable C2. The 25-pair cable from the Analog “Line” CP provides connectivity for eight voice terminals at the Line input to the ETU. Since the ETU supports only five PFT stations, three of the voice terminals are wired straight through the ETU and are not switched during service interruptions. A similar condition exists for the 25-pair cable (D) from the CO Trunk CP to the Trunk input of the ETU. Three of the eight trunk port appearances are wired straight through the ETU to the CO and are not switched. Trunk ports connected by legs 2 and 3 of the splitter cable are wired directly to the TAE Block.

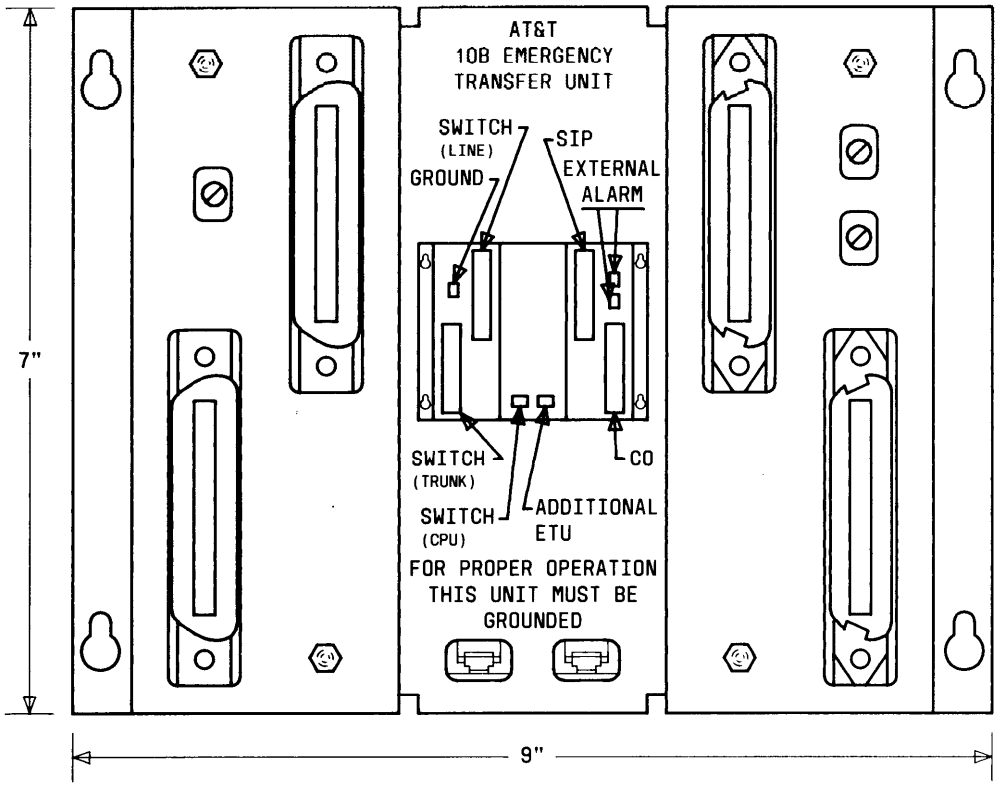
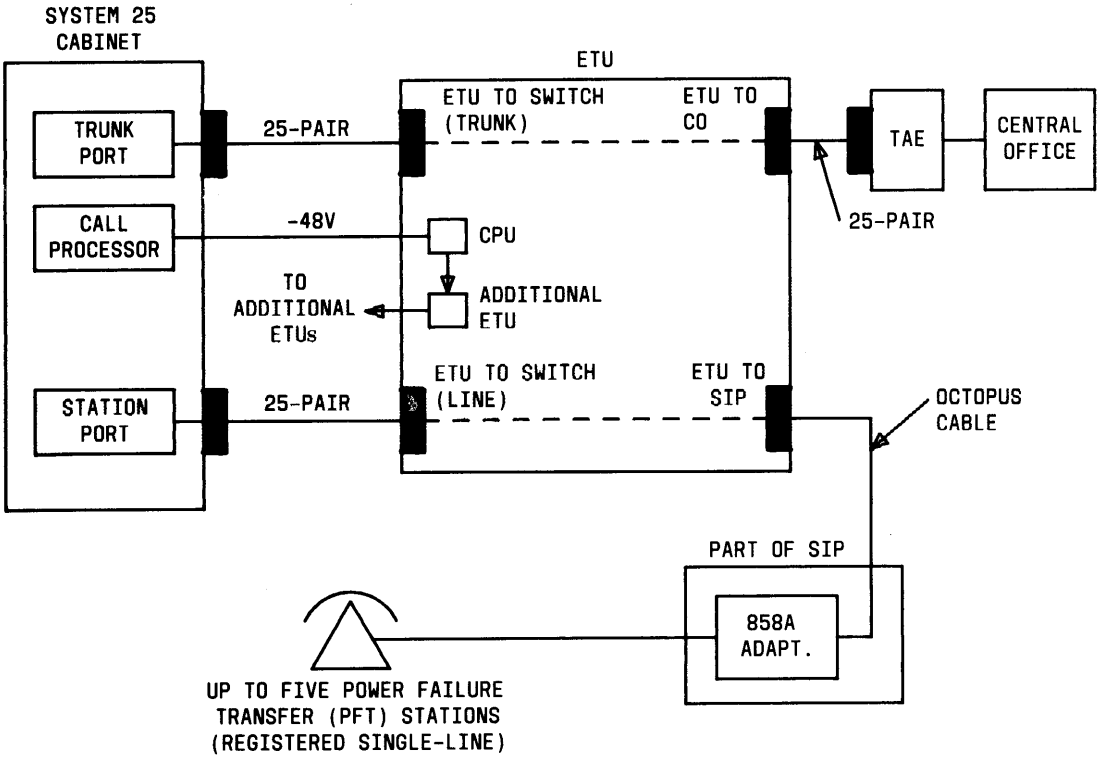
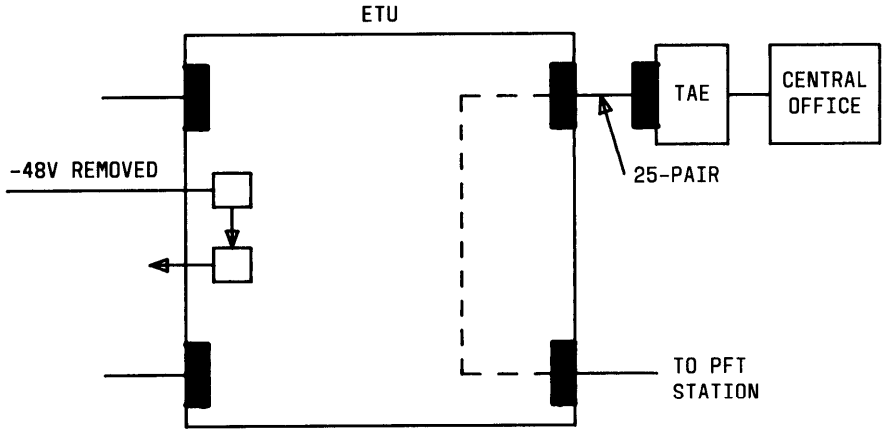


Figure 2-31. 10B Emergency Transfer Unit (ETU)

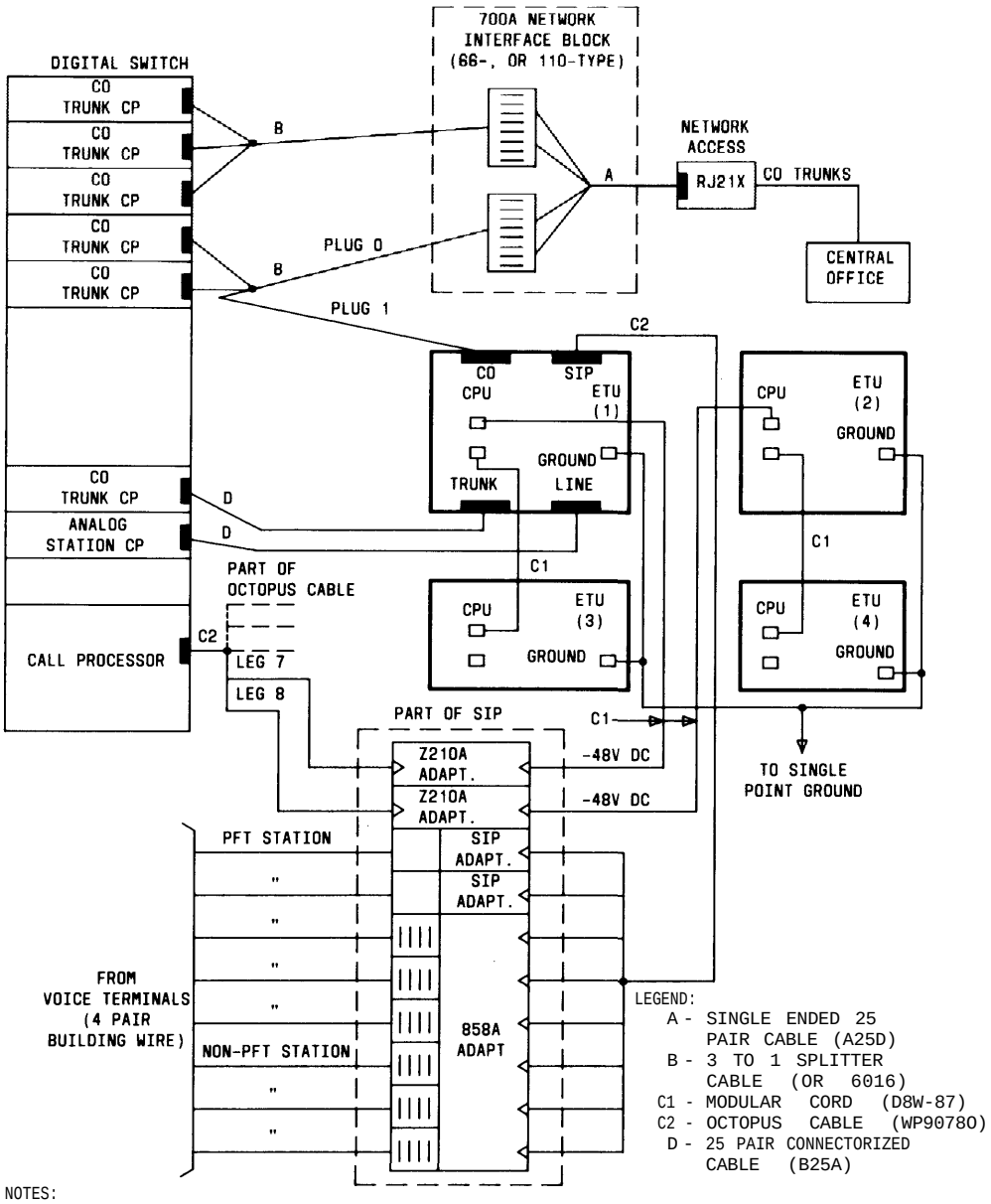


(a) ETU CONNECTIONS (DASHED LINES) NORMAL OPERATION - (NO PFT)
(TRUNK SUPPORTS STANDARD CO CALLS)



(b) ETU CONNECTIONS (DASHED LINES) ON PFT

Figure 2-32. Emergency Transfer Unit Connections



NOTES:
1. TRUNK AND STATION CONNECTIONS TO ETU 2-4 ARE SIMILAR TO ETU (1).
2. THREE OF EIGHT STATION LINES (FROM ANALOG STATION CP) AND THREE OF EIGHT TRUNKS (FROM CO TRUNK CP) WHEN USED ARE "FED-THROUGH" ETU, THEY ARE NOT SWITCHED IN THE EVENT OF A POWER FAILURE.
3. MAXIMUM ETU(S) = 4, MAXIMUM PFT STATIONS PER ETU = 5.

Figure 2-33. Multiple ETU Arrangements

Program

Description

This feature enables system users to store numbers for access by feature buttons or code dialing. Multiline voice terminal users can program numbers on REP DIAL and FLEX DSS buttons. Both multiline and single-line terminal users can store Personal Speed Dialing numbers. FLEX DSS buttons provide access to inside extension numbers only; REP DIAL buttons and Personal Speed Dialing are used for account codes and outside numbers (maximum of 28 digits and 25 digits, respectively).

Special Characters

The following special characters may be used in Repertory Dialing and Personal Speed Dialing numbers:

CHAR.	USED IN REPERTORY DIALING NUMBERS
*	Produces a 1.5 second pause. — Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent. — The pause character should not be programmed for internal calls.
#*	Transmits an actual "*".
# #	Transmits an actual " #".
#1xx	Represents a System Speed Dialing code (where xx = 00-89) or a Virtual Facility code (where xx = 90-99). If using a Virtual Facility code, it may appear only at the beginning of the stored number.
#2x	Represents a Personal Speed Dialing code (where x = 0-9).
#3x	Represents additional Personal Speed Dialing codes (where x = 0-9).
#8	Marks the beginning of End-to-End Signaling. (System begins transmitting touch-tone signals to the far end switch.)

CHAR.	USED IN PERSONAL SPEED DIALING NUMBERS
*	Produces a 1.5 second pause. — Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent. — The pause character should not be programmed for internal calls.
#*	Transmits an actual "*".
# #	Transmits an actual "#".
#1xx	Represents a System Speed Dialing code (where xx = 00-89) or a Virtual Facility code (where xx = 90-99). If using a Virtual Facility code, it may appear only at the beginning of the stored number.
#8	Marks the beginning of End-to-End Signaling. (System begins transmitting touch-tone signals to the far end switch.)

Programming Procedures

Program mode may be entered either of the following methods:

- At any voice terminal, by dialing the programming access code (#4).
- At terminals equipped with a Test/Program switch, by moving the switch to position P; this method cannot be used at Switched Loop Attendant Consoles (SLACs).

If the code is used to enter program mode, the terminal remains in program mode until the user goes on-hook or a timeout occurs. If the switch is used to enter program mode, the terminal remains in program mode until the switch is returned to the midpoint between P and T; the system will send a single-ring reminder every 60 seconds until the switch is repositioned.

Voice terminals equipped with a display enhance theprogramming procedure by displaying a prompt on the screen and then the digits as theyare dialed. Refer to "Interactions" for additional information.

Programming a number always removes the number that was previously stored in the same location. If a user wants to remove an old number and *not* replace it with a new one, Step 3 in both of the following procedures should be skipped.

To Program a Number by Using the Code:

1. Lift the handset and listen for dial tone.
2. Dial #4 to enter the Program mode.
3. Dial the number you want to program.
4. Either:

   — Press the FLEX DSS or REP DIAL button,

   or

   — Dial the Personal Speed Dialing code (#20-#39) to indicate where this number should be stored.
5. Listen for confirmation tone and dial tone.
6. To program another number immediately, repeat steps 3 through 5.
7. Hang up.

To Program a Number by Using the Switch (non-SLAC terminals only):

1. Slide the switch on the left side of the voice terminal to position P.
2. Lift the handset and listen for dial tone.
3. Dial the number you want to program.
4. Either:

   — Press the FLEX DSS or REP DIAL button,

   or

   — Dial the Personal Speed Dialing code (#20-#39) to indicate where this number should be stored.
5. Listen for confirmation tone and dial tone.
6. To program another number immediately, repeat steps 3 through 5.
7. Hang up.
8. Slide the switch back to the midpoint between P and T.

Considerations

The Program feature allows users to assign their own Personal Speed Dialing/Flex DSS/Repertory Dialing numbers. This is particularly useful where working arrangements or personnel responsibilities change.

Users cannot place or answer calls while in the program mode. Incoming calls will receive Busy Tone.

In the unlikely event that a number contains more digits than are free in the common Personal Speed Dialing/Repertory Dialing memory (approximately 34100), reorder tone will be returned after the indication of where this number is to be stored (see Procedures, above).

Interactions

The following features interact with Program.

Bridging of System Access Buttons: A bridging station user can program only the Personal Speed Dialing codes, REP DIAL buttons, and FLEX DSS buttons associated with the bridging station, not with the principal station.

Display: When the user of a display-equipped voice terminal enters Program Mode, a prompt is displayed on the terminal screen.

Screen 1	PROGRAM

Once the user begins to dial digits, the prompt is removed. The dialed digits are displayed, beginning on Screen 1 and continuing on Screen 2, if necessary.

Screen 1	2653

Character position 16 on Screen 1 is reserved for queue data, which the system continues to update for SLACs and DGC terminals.

The Program display remains on screen until the user selects the button or code to be programmed. After confirmation tone is returned, the PROGRAM prompt is again displayed. The user can then go through additional programming sequences, if desired. After programming is completed, the user can go on-hook or exit from the Program Mode to Normal Mode by switchhook change (if the mode was entered by dialing #4) or by returning the PROGRAM switch to its neutral position (non-SLAC terminals).

Remote Access: Remote Access callers cannot use the Program feature.

Administration Requirements

Voice Terminal Port:

- Assign FLEX DSS and REP DIAL buttons.
- Allow Personal Speed Dialing on a per-station basis.

Recall

Description

Users of single-line voice terminals that have RECALL buttons can obtain System 25 recall dial tone by pressing the button. Pressing RECALL is equivalent to briefly pressing and releasing the switchhook (switchhook flash), which is the required method of getting recall dial tone at a terminal not equipped with the RECALL button.

Multiline voice terminals are administered for either manual or automatic activation of the Callback Queuing feature. Operation of the RECALL button is the manual method. Refer to the “Callback Queuing” feature description for complete information. Use of the RECALL button for callback does not interfere with its Centrex functions, described in the next paragraph.

The RECALL button on a multiline voice terminal can be used, under specialized conditions, to send a switchhook flash to the Central Office (for example, to access Centrex services). However, it can never be used to send a switchhook flash to the System 25.

Interactions

The following feature interacts with Recall.

Callback Queuing: The RECALL button can be used to send switchhook flash to Centrex trunks. If a conference exists with a queuable tone and a Centrex trunk, the first push of RECALL queues the call. A second push of RECALL is needed to send switchhook flash to the Central Office.

Administration Requirements

For Centrex operation, stations must not be assigned toll restriction, and the Centrex trunks must be administered as Type 805.

Remote Access

Description

This feature allows a caller to dial into a System 25 from the public network using a predetermined 7- or 10-digit number and use some features and services. The caller may be required to dial a barrier (security) code after reaching the system to access the features and services.

System administration can assign CO, FX, or INWATS trunks for Remote Access calls. These trunks can be dedicated to Remote Access or shared with other kinds of calling. On dedicated trunks, all incoming calls receive special Remote Access treatment. On shared trunks, Remote Access is allowed only during night service times; at other times incoming calls receive standard trunk treatment.

If DID service is in use, a valid DID number (otherwise unassigned) can be used for Remote Access. The system will answer calls to this DID number and return special dial tone to the caller. From this point, operation is exactly like Remote Access via the trunk types listed above. Note that since the DID Remote Access number does not become “busy” until all DID trunks are busy, multiple Remote Access calls can be active at the same time.

When the system recognizes an incoming call as a Remote Access call, it determines whether the trunk is shared. If it is, and if night service is not active, the call can be routed to a personal line appearance or to a DGC group. If neither of these answering points is administered, the call goes to an attendant for handling.

If the trunk is dedicated to Remote Access or if night service is active, the caller gets special dial tone. If a barrier code is not required, the caller can now dial the desired number or feature access code. Calling privileges of the caller are determined by a system-wide default Class of Restriction (COR).

If a barrier code is required, the caller dials five more predetermined digits on hearing the special dial tone and receives second dial tone. After second dial tone the caller can call the desired number or feature access code. The specific barrier code used determines the calling privileges of the user. Up to 16 different barrier codes can be administered for the system.

When the Remote Access caller is using a rotary voice terminal, dial pulses are not accepted by the system after the special dial tone is returned. When dial tone times out, the call routes to an attendant; if night service is active, the call routes to an administered “backup” multiline station.

If a Remote Access call reaches an inside station, the station can transfer the call back out of the system subject to System 25 trunk-to-trunk restrictions and the transferring station’s calling restrictions. This is allowed even if the caller’s barrier code does not allow a direct call out.

Considerations

Using Remote Access, an employee of a company with a System 25 PBX can access system facilities from home or other remote locations. This is a valuable feature for salesmen on the road and people at small branch offices.

Stations receiving Remote Access calls can treat them like any other kind of incoming call and use features such as Pickup and Forwarding.

Features requiring recall dial tone (for example, Park and Transfer) cannot be used by Remote Access callers.

Interactions

The following features interact with Remote Access.

Account Code Entry: This feature cannot be used by Remote Access callers.

Call Accountability: This feature cannot be used by Remote Access callers.

Calling Restrictions: A barrier code class of restriction (COR) has the same parameters as the class of service permissions associated with stations. A system-wide default COR must be administered for use if barrier codes are disabled. Barrier code CORs override the default COR.

Conference: This feature cannot be used by Remote Access callers.

Display: Since remote access calls are all incoming trunk calls, the display at the receiving station has the standard “reception of outside calls” format. The receiving station has no special indication that this is a remote access call.

A display set user who bridges onto a Personal Line appearance where a remote access call is active will have the display updated for conference status.

Direct Group Calling (DGC): Remote Access callers cannot attempt to log in or out of a DGC group.

Following: This feature cannot be used by Remote Access callers.

Forwarding: Remote Access calls to a System 25 station that has the Forwarding feature activated will forward like any other incoming calls to the station.

Remote Access callers can activate Forwarding to outside numbers at System 25 stations. Forwarding to other inside stations is not allowed.

Hold: This feature cannot be used by Remote Access callers.

Night Service, Directed: Remote Access trunks (dedicated or shared) cannot be given Directed Night Service treatment.

Night Service, Trunk-Answer-from-Any-Station (TAAS): Remote Access trunks (dedicated or shared) cannot be given TAAS Night Service treatment. Remote Access callers cannot answer TAAS Night Service calls.

Park: This feature cannot be used by Remote Access callers.

Pickup: This feature cannot be used by Remote Access callers.

Program Mode: This feature cannot be used by Remote Access callers.

Remote Initialization and Maintenance Service (RIMS): RIMS requires a unique barrier code and carries special non-administrable restrictions.

Speed Dialing, Personal: This feature cannot be used by Remote Access callers.

Station Message Detail Recording (SMDR): Remote Access calls are included in the SMDR records. The following unique data is presented for Remote Access calls (in addition to type, data, time, and duration).

Remote Access-to-Inside Number:

- CALLED NUMBER field—IN.
- FAC field—the number of the incoming trunk.
- STN field—the dial code of the called party; if the call timed out to the attendant, the attendant's code is shown.
- ACCOUNT field—the Barrier Code number; if the call timed out to the attendant, this field is blank.

Remote Access-to-Outside Number:

- CALLED NUMBER field—the outside number dialed by the remote caller.
- FAC field—the number of the outgoing trunk or trunk pool.
- STN field—the number of the incoming trunk.
- ACCOUNT field—the Barrier Code number.

Transfer: This feature cannot be used by Remote Access callers.

Administration Requirements

System:

- Enable barrier codes (yes or no; default = yes).
- Assign DID remote access number (PDC or 0; default = 0).

- Assign system default COR for Remote Access:
  - ARS Facility Restriction Level (0 to 3; default = 3).
  - Toll Restriction Class (1 to 4 or 0 for none; default = 0).
  - Outward Restricted? (yes or no; default = no).
  - "CO pool" dial restricted? (yes or no; default = no).
  - "Other pools" dial restricted? (yes or no; default = no).
- For each barrier code defined:
  - Barrier code number (1 to 16).
  - Barrier code digits (five characters or enter 0 to remove; valid characters are 0 to 9, * and #).
  - Assign barrier code's COR:

    ARS Facility Restriction Level (0 to 3; default = 3).

    Toll Restriction Class (1 to 4 or 0 for none; default = 0).

    Outward Restricted? (yes or no; default = no).

    "CO pool" dial restricted? (yes or no; default = no)

    "Other pools" dial restricted? (yes or no; default = no).
- Assign Remote Access Night Service "backup" multiline station (PDC or 0; default = 0).
- Assign RIMS barrier code digits (5 digits [0 to 9, * #] or 0 to remove; default = 98765)

Trunk Port:

- Specify Remote Access usage (no, shared, or dedicated; default = no).

Remote Administration Interface

Description

This feature provides dial-up access to the system's administration port, either for a standard system administration terminal or for a PC running Advanced Administration software.

Both read and write capability is provided with access to all system translation and fault tables. A remote administration terminal can perform the same functions as the on-premises SAT.

Remote Administration allows remote access to the system by maintenance personnel, the System Administrator, and others.

Interactions

Only one System Administration Terminal can be connected at one time.

Administration Requirements

Depends on the connecting arrangements selected (see below).

Hardware Requirements

Requires a remote SAT.

Requires that port #1 of the CPU/MEM CP be connected to: (1) a dedicated modem and dedicated facility (private line or CO trunk), or (2) a dedicated modem connected to a tip ring station port, or (3) an ADU connected to a data line port. See the "System Administration" feature description for additional information. Connectivity information is also provided.

Remote Initialization and Maintenance Service (RIMS)

Description

Remote Initialization and Maintenance Service (RIMS) provides an AT&T technician remote access to System 25. Using the RIMS feature, the technician can do System 25 initialization, ongoing administration, and maintenance.

The remote administrator can initialize translations after the system is installed. Unless a hardware change is required, the remote administrator can do ongoing administration without having to visit the customer's site. Similarly, to troubleshoot a problem, a technician can call the RIMS port and check the Error Log to determine the probable cause of the trouble. The technician can clear alarms remotely and decide whether a service dispatch is necessary.

Considerations

RIMS may not be available in some areas of the country.

Interactions

A call to the RIMS port is logged by SMDR.

Administration Requirements

For a RIMS call to be made during initialization, at least one voice station and one trunk must be translated. A "full-default" cold start provides these translations.

Repertory Dialing

Description

This feature allows multiline voice terminal users to store a telephone number, account code, or feature access code in the system's memory and associate that number with a REP DIAL button. Pressing REP DIAL is equivalent to dialing the stored number. Individual numbers can be up to 28 digits in length.

Programming the number is accomplished from the user's voice terminal. Programming procedures and other information can be found in the "Program" feature description. Should the user attempt to enter more than 28 digits, Reorder Tone will be given.

The user can press REP DIAL under any of the following conditions:

1. When off-hook receiving Dial Tone
2. When off-hook on a call on which more dialed digits are expected
3. When off-hook on a call and connected to an outgoing trunk (End-to-End Signaling might apply in this case)
4. After pressing ACCT ENTRY or dialing the Account Code Entry access code.

When REP DIAL is pressed, the button's status LED lights briefly and then goes dark.

Considerations

Repertory Dialing simplifies dialing long or frequently called numbers, and allows one-button access to many features.

Interactions

The following features interact with Repertory Dialing.

Account Code Entry: An Account Code may be stored on a REP DIAL button. The REP DIAL button should be pressed at the point where the account code would normally be dialed.

Bridging of System Access Buttons: If a station user selects a bridged appearance for an outgoing call and then presses a REP DIAL button, the digits programmed into the button are outpulsed as they would be if the user had selected one of the station's own System Access buttons.

Calling Restrictions: A user can not use Repertory Dialing to access a number that he or she is restricted from dialing.

Display: When a call is placed by pressing a REP DIAL button, the characters stored are displayed. If * was programmed to store a pause, P is displayed in the position of the *. If * was stored by programming #*, only * is displayed. If the button is not programmed, REP DIAL is displayed.

Following/Forwarding: The associated activation and deactivation sequences (or portions of them) can be stored on REP DIAL buttons.

Last Number Dialed: A number called by pressing a REP DIAL button is saved by the Last Number Dialed feature.

Speed Dialing: Numbers already stored as System Speed Dialing numbers can also be stored as Repertory Dialing numbers. Storing a System Speed Dialing code (#100-#189) on a REP DIAL button saves memory space (compared to storing the whole number again on a REP DIAL button).

Virtual Facilities: Virtual Facility codes can be stored on REP DIAL buttons.

Voice Terminal Port:

- Assign Repertory Dialing (REP DIAL) buttons.

Send All Calls

Description

This feature allows multiline voice terminal users to turn off their ringers and invoke a “do not disturb” condition toward incoming calls. In addition, users who have coverage or bridged appearances will have those calls directed immediately to their covering and/or bridging stations, without the normal system ringing delay. Send All Calls also allows covering users to temporarily remove their voice terminals from the coverage path.

This feature is activated by pressing the SEND ALL CALLS button. It is deactivated by pressing the button a second time.

Considerations

Send All Calls gives the user the option of having incoming calls sent directly to coverage or making the terminal busy to incoming calls without sending them to coverage. The feature is intended for occasional or temporary use.

Send All Calls must be assigned to a button that has a status light. The light turns on when the feature is in effect.

The following types of calls always ring at a station, regardless of the status of Send All Calls:

- Automatic Intercom calls.
- Directed Night Service calls.
- Calls to a PDC that is signed in at the station.

  Note: Calls to an FPDC do not ring when Send All Calls is in effect.

- Calls returning to a DTAC on RTN-BUSY or RTN-DA buttons.

Send All Calls cannot be assigned to a SLAC. The Attendant Position Busy feature provides a similar capability.

When Send All Calls is in effect at a station and incoming calls are directed to coverage, ringing at the sending station is not necessarily canceled completely. A single-ring reminder for incoming calls is optional, assigned by the System Administrator for each Send All Calls button.

On calls to non-busy stations where Send All Calls has been activated, the callers hear ringing until a covering station answers, or, if the station is not covered, until the call is dropped.

Interactions

The following features interact with Send All Calls.

Bridging of System Access Buttons: The principal station can be administered so that pressing the SEND ALL CALLS button will send ringing for incoming calls to its coverage stations only, to its bridging stations only, or to both.

If ringing is sent to a Bridged Access (BA) button via Send All Calls, and if the BA button is administered to not receive ringing, the call will flash (but not ring) at the BA button. If ringing is sent to a BA button via Send All Calls, and if the BA button is administered to receive ringing (immediate or delayed), then the call will ring *immediately* on the BA button.

Callback Queuing: Callback attempts to the originator are not affected by Send All Calls.

Call Waiting: A busy station with Send All Calls activated will receive call waiting tones; the caller will hear special ringback. If the busy station then goes on-hook, single-ring reminder will not be given for that waiting call.

Coverage (General): Send All Calls works in conjunction with the Coverage features at covered and covering stations. At stations not associated with Coverage, Send All Calls simply serves to silence the ringer on incoming calls; no redirection occurs.

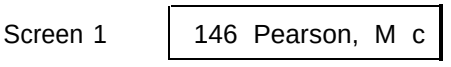
If a station is translated to *not* send ringing to coverage when calls to this station are unanswered, the Send All Calls feature overrides this instruction (ringing *will* be sent).

If a covering station activates Send All Calls, the station is removed from the coverage path completely. Coverage calls will not be directed to the station.

Coverage, Group: If a station with Send All Calls activated has group coverage and all the coverage receivers are busy, a call waits at the station while the system periodically checks for an idle receiver. When one becomes available, the call is directed to the covering station.

Coverage, Individual: If a station with Send All Calls activated has only individual coverage and all coverage receivers are busy, a call stays at the station; it does not go to coverage.

Display: When Send All Calls is invoked, the sending station still receives incoming call information. If the calls are being sent to coverage, the proceeding-to-coverage descriptor “c” appears in position 16.



Distinctive Ringing: Normal audible ringing is turned off for incoming calls when Send All Calls is activated, unless single-ring reminder is administered.

Forwarding: Forwarding supersedes Send All Calls. A call forwarded from a station with Send All Calls activated will not go to coverage or to bridging stations unless the call is not answered at the forwarded-to station and returns. After returning, the call routes according to the Send All Calls feature.

Hands-Free Answer on Intercom: Activating Send All Calls will cause an active AUTO ANS button to turn off. As long as the Send All Calls feature is in use, AUTO ANS cannot be turned cm.

Line Status Indications: The line status lights still flash for incoming calls when Send All Calls is in effect even though normal ringing is cut off. The lights stop flashing when the calls are answered by a covering station or dropped by the caller.

Personal Line: Ringing on Personal Lines is turned off by activation of Send AH Calls whether the station is the principal (owner) of the line or not. Personal Line calls follow the coverage arrangements of the principal station. If the principal station is not covered, the call will simply stay at the principal station until dropped (even if other stations with that Personal Line have coverage).

Administration Requirements

Voice Terminal Port:

- Assign Send All Calls button.
- Assign single-ring reminder if desired.
- Send ring to bridged stations only, to coverage stations only, or to both?

Speaker

Description

Some 7300 H-series voice terminals have a built-in loudspeaker that allows on-hook dialing, group listening, and monitoring of call progress signals. The terminal user turns on the speaker by pressing the SPEAKER button. Pressing the button at an idle terminal has the same effect as lifting the handset: the user is connected to the selected line and hears Dial Tone. An associated LED is lighted when the Speaker is on. Speaker volume may be adjusted by the terminal's volume control located on the left side of the set.

The speaker and associated LED are turned off by pressing SPEAKER again or by lifting the handset. The latter operation connects the handset to the associated voice channel. When using the handset, pressing SPEAKER will turn on the speaker to support the Group Listen feature; pressing SPEAKER again will turn off the speaker and associated LED. Note that once the user has lifted the handset, it is possible to return to "hands-free" operation only by putting the call on hold, hanging up the handset, then reconnecting the call by pressing SPEAKER. Hanging up the handset will terminate the call whether the speaker is on or off.

Note: The built-in speaker provides one-way communication (listen only). The user must pick up the handset to converse.

Considerations

The built-in speaker supports group listening, monitoring of calls (e.g., while waiting on hold), and on-hook dialing.

Hardware Requirements

Only 7300H-series (MERLIN) voice terminals with a SPEAKER button support this feature. Sets with a SPEAKERPHONE button have full speakerphone service, which provides two-way, on-hook calling.

Speakerphone Adjunct

Description

The speakerphone adjunct permits users of voice terminals not equipped with built-in speakerphones to place and receive calls without lifting their handsets. The adjunct has an On/Off switch, a switch to temporarily mute the microphone, status lamps, and a volume control (for incoming voice only).

Ail voice terminal features operate normally with the speakerphone adjunct.

Lifting the handset during speakerphone operation automatically turns off the speakerphone. The speakerphone may be turned on during a call by pressing the On/Off switch and hanging up the handset.

Considerations

Speakerphone operation allows users to perform other activities while carrying on a conversation. Speakerphones also facilitate conference calls.

Interactions

The following feature interacts with Speakerphone Adjunct.

Headset Adapter Adjunct: A voice terminal cannot have both a speakerphone adjunct and a headset adapter adjunct.

Hardware Requirements

4A Speakerphone System

The 2500SM single-line voice terminal and 2991-type 10-Button MET set require a 4A Speakerphone System. The 4A (Figure 2-34) provides a speaker and associated microphone, indicator lamp and operating controls. The controls include a two position ON OR QUIET/OFF rocker switch and a volume control.

The 4A Speakerphone requires an 85B1 power unit.

S1O1A/S102A Speakerphone (PEC 3163-HFU)

The S101A Speakerphone (Figure 2-34) is used with the 12-Button MET Set (7203 M). The S102A Speakerphone is used with 7300 H-series voice terminals except the 5-Button and HFAI sets.

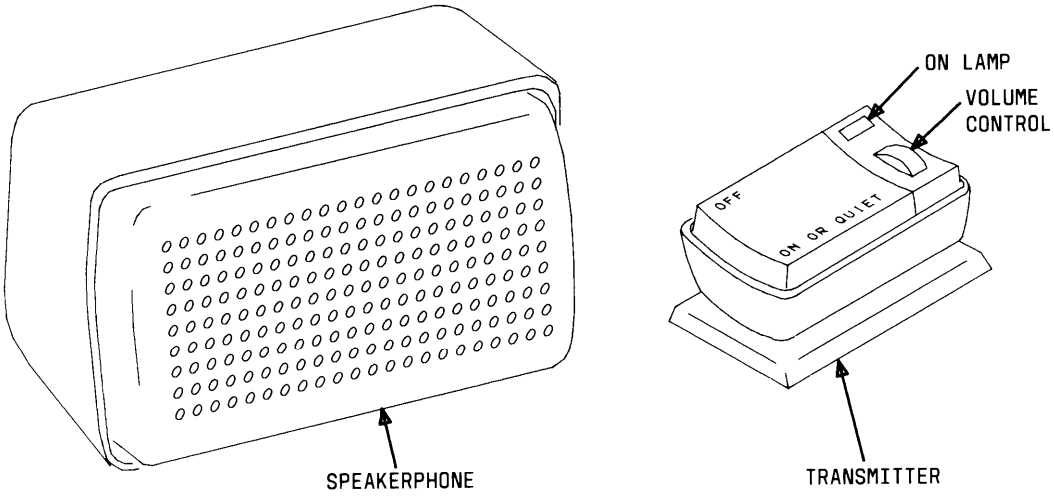
The S101A/S102A speakerphones are equipped with a 4-foot connecting cord that plugs into the voice terminal. Connecting cords are available in optional lengths of 18 inches and 14 feet.

The unit has a SPEAKERPHONE ON/OFF pushbutton switch and a MICROPHONE ON/OFF pushbutton switch. The former controls the entire unit; the latter turns the microphone on and off for privacy. Each button has an associated green status LED.

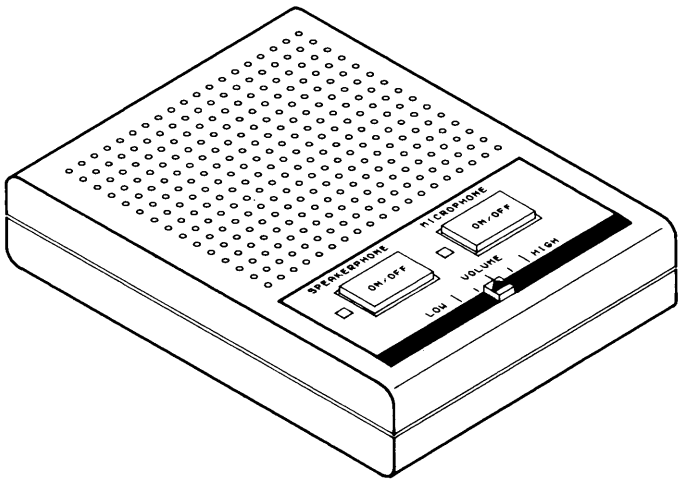
The S101A Speakerphone must be powered locally with a 2012D Transformer that plugs into a 115V ac receptacle. Adjunct power supplies are described in Section 4, “Hardware Description.” The S102A Speakerphone does not require supplemental power, except when used with a 34-Button Deluxe voice terminal.

Detailed speakerphone adjunct connection information is provided in the following figures:

- Figure 2-35—Speakerphone Connections for 7300H Series Multiline Voice Terminals (Except 34-Button Sets)
- Figure 2-36—Speakerphone Connections for 34-Button Voice Terminals
- Figure 2-37—Speakerphone Connections for 12-Button MET Sets.

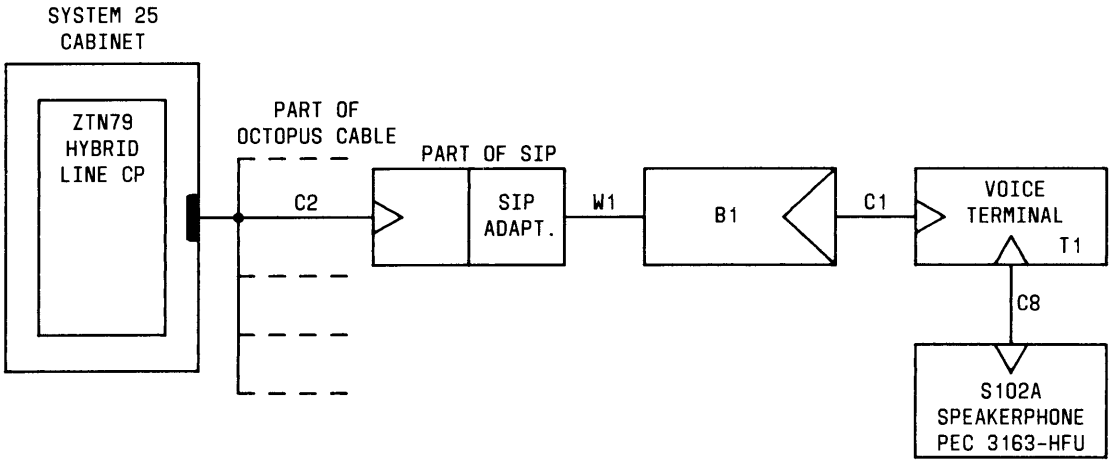


4A SPEAKERPHONE SYSTEM



S101A/S102A SPEAKERPHONE

Figure 2-34. Speakerphone Adjuncts

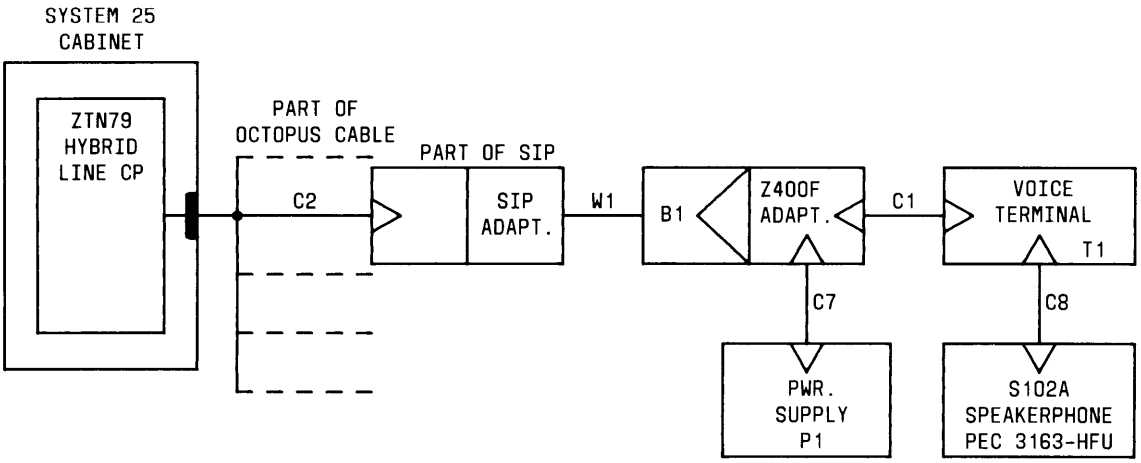


LEGEND:

B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
T1 - 7300H SERIES VOICE TERMINALS EXCEPT 34-BUTTON DELUXE
W1 - 4-PAIR INSIDE WIRING CABLE*

* - FURNISHED BY INSTALLER

Figure 2-35. Speakerphone Connections For 7300H Series Multiline Voice Terminals (Except 34-Button Sets)

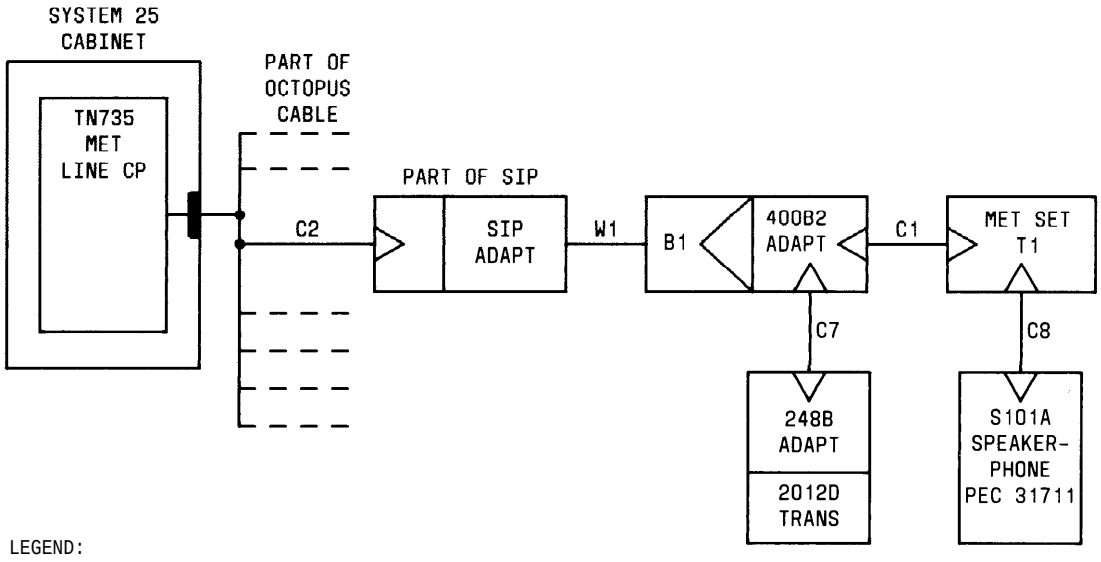


LEGEND:

B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
T1 - 7305H02B VOICE TERMINAL (34-BUTTON DELUXE)
W1 - 4-PAIR INSIDE WIRING CABLE*
C7 - MODULAR CORD (D6AP-87)
PI - KS-22911 POWER SUPPLY
Z400F - ADAPTER
} PEC 62510

* - FURNISHED BY INSTALLER

Figure 2-36. Speakerphone Connections For 34-Button Voice Terminals



LEGEND:

B1 – TYPICAL-103A CONNECTING BLOCK*
C1 – MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 – OCTOPUS CABLE (WP90780) PEC 2720-05P
C8 – SPECIAL CORD - FURNISHED WITH ADJUNCT
T1 – 7203M SET - 12-BUTTON MET SET
W1 – 4 PAIR INSIDE WIRING CABLE*

248B ADAPTER – MODULARIZES 2012D TRANSFORMER
400B2 ADAPTER – POWER ADAPTER
2012D TRANSFORMER – 15-18V AC SOURCE
C7 – MODULAR CORD (D6AP-87)

} PEC 21691

* – FURNISHED BY INSTALLER

Figure 2-37. Speakerphone Connections For 12-Button MET Sets

Speed Dialing

Description

There are two types of Speed Dialing: (1) System Speed Dialing, and (2) Personal Speed Dialing.

System Speed Dialing:

Allows the System Administrator to store up to 90 numbers (maximum of 28 characters in length) that are accessible by dialing 3-digit codes from any voice or data terminal.

Examples of typical System Speed Dialing numbers include frequently-dialed DDD numbers (together with leading facility access codes for WATS, FX etc.) and account codes.

The following special characters may be used in System Speed Dialing numbers.

CHAR.	FUNCTION
*	Produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)
#*	Transmits an actual "*".
# #	Transmits an actual " # ".
#1xx	Represents a Virtual Facility code (where xx = 90-99). This may appear *only* at the beginning of the stored number.
#8	Marks the beginning of End-to-End Signaling. (System begins transmitting touch-tone signals to the far end switch.)

A user cannot use a Speed Dialing number for which he/she is toll restricted, outward restricted, or facility access restricted.

System Speed Dialing Operation: To place a call using a System Speed Dialing number, the user goes off-hook and presses the # button on the dial pad followed by the 3 digit code assigned to the desired number. The system interprets the associated stored number as if it were dialed directly. This includes analysis of the number for the various types of restriction.

This feature can also be used when entering account codes. After pressing ACCT ENTRY or dialing "*0", the user can enter a System Speed Dialing code. The stored number associated with the code (the account code) will be listed in the SMDR report.

Multiline voice terminals users may assign System Speed Dialing codes to Repertory Dialing (REP DIAL) buttons.

Personal Speed Dialing:

Allows users to program up to twenty Personal Speed Dialing numbers (maximum of 25 characters in length) that are accessible only from their terminals. The numbers are accessed by dialing associated access codes (#20-#39).

Personal Speed Dialing is authorized on a per-station basis through System Administration. The System Administrator will inform users if they can use this feature.

Refer to the "Program" feature description for more information about programming Personal Speed Dialing numbers.

If enough storage space is available in memory to allow assignment of a Personal Speed Dialing number, confirmation tone will be returned after each number is programmed. If not, reorder tone will be returned.

Note: Personal Speed Dialing is voice terminal oriented, not PDC oriented. A user who logs in at another terminal cannot use his/her Personal Speed Dialing numbers.

Considerations

System Speed Dialing allows users to dial a number by simply dialing #100-#189. The stored number associated with each code is (typically) a common-use phone number and is programmed via System Administration.

Personal Speed Dialing allows users to program up to twenty numbers for their personal use; these numbers can only be accessed from the terminal where originally programmed.

The system will compare the restrictions applicable for the voice terminal against the number associated with the Speed Dialing code, then allow or deny the call just as if the number had been dialed directly from the terminal.

Interactions

The following features interact with Speed Dialing.

Account Code Entry: Speed Dialing codes may be used to store account codes.

Bridging of System Access Buttons: Personal Speed Dialing is a station oriented feature. If a station dials a Personal Speed Dialing code (#20-#39) while off-hook on a Bridged Access button, the system will handle this call exactly as if the code was dialed from this station's System Access button.

Calling Restrictions: A terminal that is restricted from placing a particular call cannot avoid restriction by using the Speed Dialing feature.

Data Terminal Dialing: System Speed Dialing codes can be entered during Data Terminal Dialing. Personal Speed Dialing is not supported.

Display: When a call is placed by dialing a Personal Speed Dialing code, the characters stored are displayed. If * was programmed to store a pause, P is displayed in the position of the *. If * was stored by programming #*, only * is displayed.

When a call is placed using a System Speed Dialing code, only the dialed code (#100-#189) is displayed.

Last Number Dialed: A number called with a Speed Dialing code is saved by the Last Number Dialed feature.

Personal Lines: The Speed Dialing feature is not accessible from Personal Lines.

Remote Access: Remote Access callers cannot use the Speed Dialing feature.

Repertory Dialing: Storing a System Speed Dialing code (#100-#189) on a REP DIAL button saves memory space, compared to storing the whole number again on the REP DIAL button.

Speed Dialing: A Personal Speed Dialing number can include a System Speed Dialing code only as the first four characters (but nowhere else). Personal Speed Dialing numbers *cannot* include Personal Speed Dialing codes. System Speed Dialing numbers *cannot* include any Speed Dialing codes.

Virtual Facilities: A Virtual Facility code may be used within Personal or System Speed Dialing numbers. When used, it must appear at the beginning of the stored number (first four characters).

Administration Requirements

System:

- Assign System Speed Dialing Numbers.

Voice Terminal Port:

- Allow/Deny Personal Speed Dialing on a per station basis.

STARLAN NETWORK Access

Description

The AT&T STARLAN NETWORK (STARLAN NETWORK) Access feature provides connectivity between System 25 and a *colocated* STARLAN NETWORK. This connectivity is provided by a combination of hardware and software elements. **The STARLAN NETWORK must use Release 2 software; System 25 is not compatible with Release 3 STARLAN NETWORK software.**

The STARLAN INTERFACE circuit pack (ZTN84) is the principal hardware element connecting System 25 and the STARLAN NETWORK. One or more of these circuit packs may be mounted in the System 25 cabinet(s). The STAR LAN circuit pack (CP) communicates with System 25 call processing over System 25's Time Division Multiplex (TDM) bus. To System 25, this circuit pack functions like a 4-port Data Line circuit pack (DLC). To the STARLAN NETWORK, the STARLAN CP appears as a STARLAN NETWORK workstation.

Communication between STARLAN NETWORK equipment (workstations, servers, hosts) and data terminals, PC6300s, and host computers connected to System 25 is provided by firmware on the STARLAN CP and communications program(s) on the PCs and hosts. Two communications programs are available to users:

- **System 25 STARLAN NETWORK ACCESS (ACCESS)**

  ACCESS allows MS-DOS personal computers (PCs) connected (via the PC's serial port) to System 25 to communicate with DOS Servers on the STARLAN NETWORK and to function as client workstations. The interface from the System 25 to the STARLAN NETWORK is the STARLAN CP operating in *bridge mode.* Bridge mode provides a transparent connection between the PC and the STARLAN NETWORK. Personal computer users may access the STARLAN NETWORK just as though they were connected to the STARLAN NETWORK with a Network Access Unit (NAU), although at lower speed. (The NAU is a CP mounted in STARLAN NETWORK workstations that permits access to other workstations and/or servers in the network.) Data transmission through the STARLAN CP is limited to a maximum of 9,600 bps. This is much less than the 1 million bps transmission rate between workstations/servers on a STAR LAN NETWORK.

  Applications that require frequent and lengthy transfers of data over the Local Area Network (LAN) will appear slow. Applications should be designed/configured to run the executable program locally (on the PC) and to access data from the file server on the LAN. ACCESS is recommended primarily for shared file and printer access. Applications should be copied to the user's (local) disk before they are run.

  This program also permits STARLAN NETWORK access for remote PCs if the System 25 is equipped with a Pooled Modem CP (TN758) or external modem pool. Remote PCs can dial the STARLAN CP through a modem using either the Direct Group Calling (DGC) feature or Direct Inward Dialing (DID) trunks to obtain a connection through System 25 to the STARLAN NETWORK.

ACCESS **must** be used in conjunction with the AT&T STARLAN NETWORK Server software (Version 2.0 or later). Installation software furnished with ACCESS requires the STARLAN NETWORK client installation diskette in order to install ACCESS.

- **Communications Access Manager (CAM)**

  CAM is an MS-DOS applications program that provides an enhanced calling interface and terminal emulation for PCs connected to System 25 or a STARLAN NETWORK. This connection must be through a DLC or a STARLAN NETWORK that is, in turn, connected to System 25 by a STARLAN CP. Refer to the Communications Access Manager (CAM) Program feature description for a more detailed description of the program.

STARLAN INTERFACE Circuit Pack

The STARLAN INTERFACE CP (ZTN84) requires a single modular connection to the STARLAN NETWORK (see Figure 2-38). It provides an interface between System 25's Time Division Multiplex (TDM) bus and STARLAN NETWORK's packet switched network. The STARLAN CP provides four full-duplex data connections at speeds up to 9,600 bits per second.

The STARLAN CP operates in two modes: Gateway Mode and Bridge Mode. Gateway mode supports connections from System 25 data terminals to STAR LAN NETWORK UNIX® system hosts, or from STARLAN NETWORK UNIX system hosts or client workstations to System 25 hosts or modem pools. In Bridge Mode, the STARLAN CP passes the STARLAN NETWORK's Universal Receiver Protocol (URP) through System 25 to a local or remote PC. This is referred to as Bridge Mode and provides a through connection between PCs running ACCESS and a STARLAN NETWORK. The proper mode (Bridge or Gateway) is auto-selected by the system.

The STARLAN NETWORK View of System 25

From the STARLAN NETWORK, the STARLAN CP functions like a STARLAN NETWORK workstation equipped with a Network Access Unit (NAU). The NAU enables STAR LAN NETWORK workstations and servers to access and exchange data over the network. Plug number 1 of the STAR LAN CP octopus cable should be connected to an "IN" jack on the Network Extension Unit (NEU) (see Figures 2-39 and 2-40).

Calls from STARLAN NETWORK to System 25

A STARLAN NETWORK workstation accesses a host computer connected to System 25 (either a local host or a remote host that can be reached using the Modem Pooling feature).

1. The STARLAN NETWORK workstation loads CLIENT and NAUCOM and then CAM software (discussed in the Communications Access Manager Program feature description) and selects a directory entry for the host.

2. CAM communicates with the STARLAN CP to place the call.

3. After a connection message is received, CAM automatically switches to terminal emulation (data) mode.
4. The user may now log into and converse with the remote host.
5. To disconnect, the user selects the CAM disconnect command.

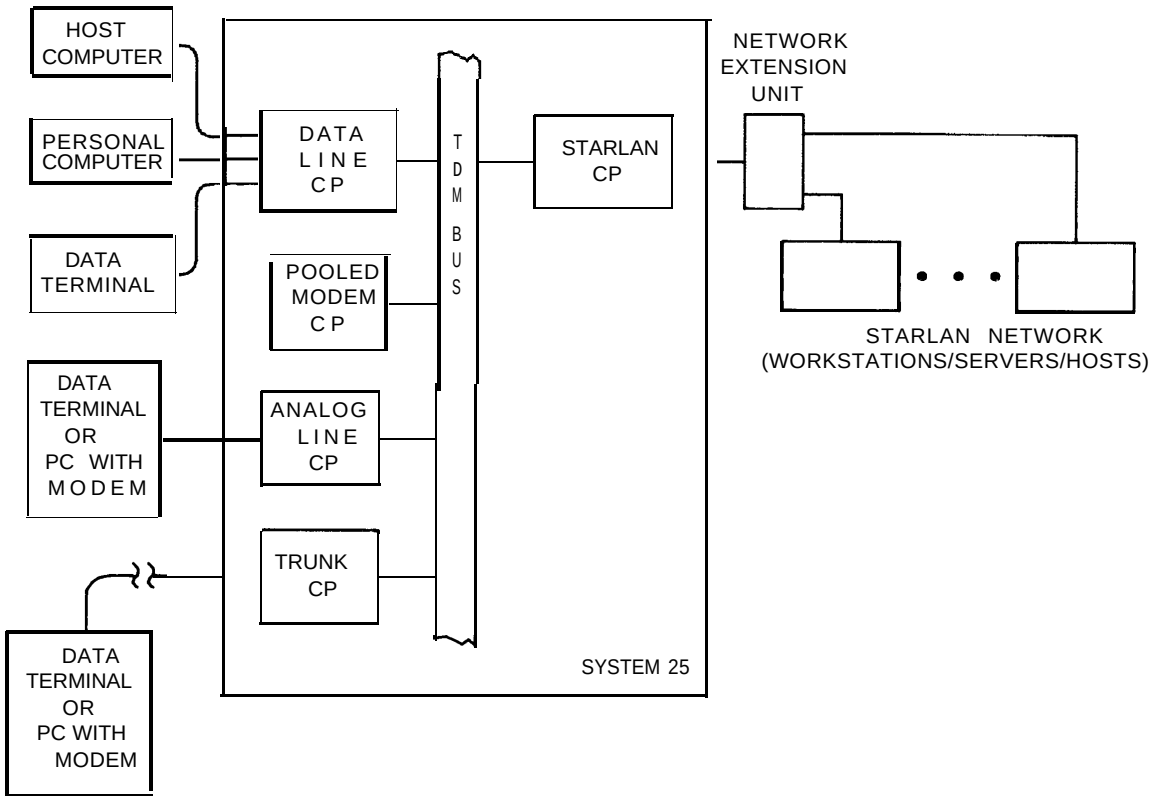


Figure 2-38. STARLAN NETWORK and System 25 Configuration

The System 25 View of a STARLAN NETWORK

From System 25, the STARLAN CP looks and functions like a TN726 DLC (with only four ports). The STARLAN CP differs from the DLC in that, when it is dialed, the STARLAN CP auto-answers the call and provides a second dialing prompt for completing the call to a STARLAN NETWORK address. Depending on user's data terminal type, the STARLAN CP automatically selects the operating mode and enables the user to access and exchange data over the network as described below.

Procedures for setting up connections (calls) between STARLAN NETWORK devices and devices connected to System 25 vary, depending on both calling and called device. Generally, a two-stage dialing procedure is used. The scenarios described below cover most situations.

Calls from System 25 to the STARLAN NETWORK

A. An MS-DOS PC connected to System 25 uses the STARLAN NETWORK ACCESS software to run STAR LAN NETWORK applications.

The PC may be connected either to a System 25 DLC port or to an analog station or trunk port. (The latter arrangement uses System 25's Modem Pooling feature.)

A typical call is as follows:

1. The PC user loads ACCESS and is automatically connected to the STARLAN NETWORK. (The STARLAN CP phone number may be entered when ACCESS is installed.)

2. The PC user may now access the STARLAN NETWORK just as if he/she were a client connected to the STAR LAN NETWORK through an NAU.

Note: Applications that are to be run frequently or are large (>10K bytes) should be copied to the user's disk before they are run.

B. A Data terminal user accesses a UNIX system host on the STARLAN NETWORK.

When the user dials the STARLAN CP, the CP answers in Gateway Mode and presents the user with a "STARLAN Address" prompt.

A typical call is as follows:

1. The user dials the STARLAN CP.

2. The STARLAN CP provides the address prompt. The user enters the logical name of the STARLAN NETWORK host (for example, 3B2).

3. The user is connected to the UNIX system host and receives the host login prompt.

Flow Control

Software flow control (XON/XOFF) may be enabled or disabled by System 25 data endpoints. After the "STARLAN Address" prompt is returned to the user, a CONTROL-X may be entered instead of a logical name. The user will be prompted further to enable or disable flow control. After that, the user is again prompted for a STARLAN address. This option also works for calls from the STARLAN NETWORK to System 25.

Data Call Disconnect

Data calls may be disconnected at either endpoint. Connections are dropped through the normal disconnect procedures of each network. If a failure in the established connection occurs, call disconnections are initiated from both sides.

Third-Party Call Setup

A data terminal (on System 25) or workstation (on the STARLAN NETWORK) can set up a call between two other stations (voice or data) using the Third-Party Call Setup feature. Since voice port/data port associations are not meaningful for STARLAN CP ports, STARLAN NETWORK workstations must always specify the Personal Dial Code of the source voice terminal or the Data Dial Code of the source data terminal. Note that this feature can only be administered for the STARLAN CP ports as a group, and not for individual STARLAN NETWORK workstations.

When placing voice calls using CAM, Third-Party Call Setup is used automatically.

Wiring

The STARLAN NETWORK wiring plan is based on standard 4-pair building wiring. The STARLAN NETWORK uses two pairs of the 4-pair cable, allowing the remaining two pairs to be used for voice service. STARLAN NETWORK data is transmitted over pairs two and three. Figures 2-39 and 2-40 provide typical connection information.

A Y-adapter may be used to combine/split the pairs at the System 25 cross-connect field. STARLAN NETWORK NAUS provide an RJ11 phone jack that terminates pair 1. Single line sets may be plugged directly into this jack (Figure 2-39). MERLIN Communications System sets require an ATL adapter and local power (Figure 2-40).

ATL Adapters

The ATL adapter (KS23475) is a connection block that provides: 48V dc power from T1 (via modular cords C4 and C1) to the ATL phone, data connections from the STARLAN workstation, and phone connections from the ATL phone (via modular cord Cl) over a shared common cable (C3) to System 25. The phone and data connections are on separate wire pairs that are split-out at the SIP “Y” adapter (Al). Phone wiring is cabled to the ATL Line CP (ZTN79) by octopus cable C2, and data wiring is cabled to the NEU by modular cord C1..

Administration Requirements

The STARLAN CP is administered as a type of data port. Some items administered on one port are automatically administered for all four ports on the CP, others are individually administrable.

Individually Administrable (default):

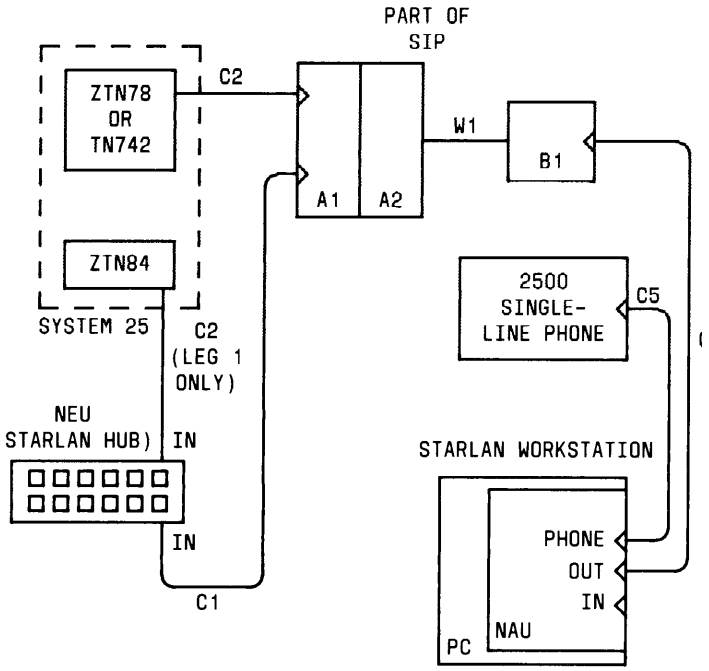
- DDC of port
- DDC to hunt to next (none)
- Display ID

Common Administration (default):

- CO trunk pool dial restriction (no)
- Other trunk pool dial restriction (no)
- Outward restriction (no)
- Toll Restriction Class (none)
- ARS FRL (3)
- Restrict Third-Party Call Setup feature (yes)

Hardware Requirements

Requires a STARLAN INTERFACE CP. Each CP provides four interface ports between the System 25 and the STARLAN NETWORK.

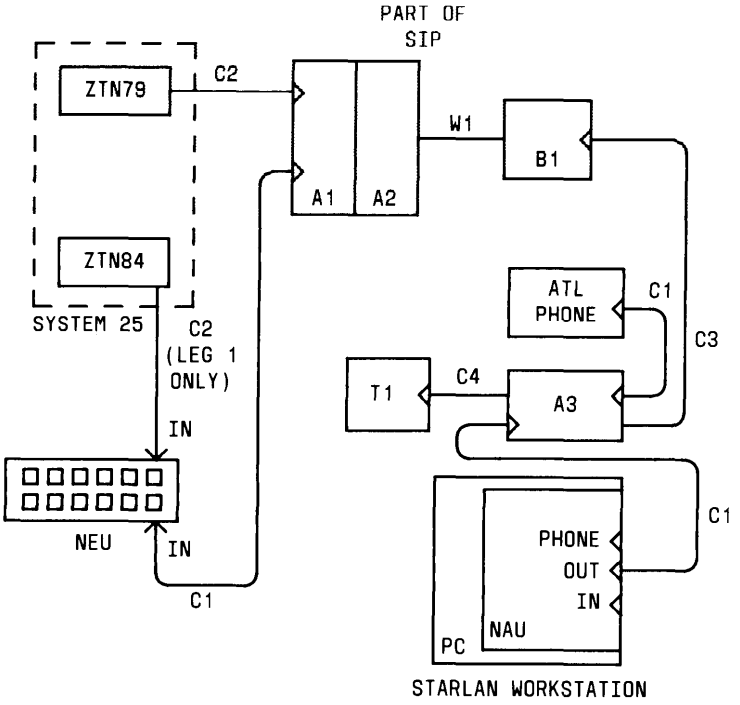


NAU - NETWORK ACCESS UNIT - PEC 2614-100
NEU - NETWORK EXTENSION UNIT - PEC 2610-001
ZTN84 - STARLAN CP - PEC 62518
ZTN78 - TIP/RING LINE CP - PEC 62504
TN742 - ANALOG LINE CP - PEC 63511
A1 - WP90851-L1 (Y ADAPTER) - PEC 2750-T05 (NOTE 1)
A2 - 858A ADAPTER
B1 - 103A CONNECTING BLOCK*
C1 - MODULAR CORD D8W-87 (FURNISHED WITH NAU)
C2 - OCTOPUS CABLE WP90780 - PEC 2720-05P (NOTE 1)
C5 - MODULAR CORD D4BU-87 (FURNISHED WITH PHONE)
W1 - FOUR PAIR BUILDING WIRING*

NOTE 1: C2 AND A1 ARE NOT REQUIRED IF NO PHONE IS PLUGGED INTO THE NAU.

* FURNISHED BY INSTALLER

Figure 2-39. STARLAN NETWORK Connection to System 25 (With 2500 Single-Line Telephone)



NAU - NETWORK ACCESS UNIT - PEC 2614-100
NEU - NETWORK EXTENSION UNIT - PEC 2610-001
ZTN79 - ATL LINE CP - PEC 62505
ZTN84 - STARLAN CP - PEC 62518
A1 - "Y" ADAPTER WP90851-L1 - PEC 2750-T05 (NOTE 1)
A2 - 858A ADAPTER
A3 - ATL ADAPTER (NOTE 1)
B1 - CONNECTING BLOCK 103A*
C1 - MODULAR CORD D8W-87 (FURNISHED WITH NAU AND PHONE)
C2 - OCTOPUS CABLE WP90780 - PEC 2720-05P
C3 - 6 INCH MODULAR CORD (PART OF A3)
C4 - 7 FOOT MODULAR CORD (PART OF A3)
T1 - 48 VOLT DC POWER SUPPLY KS22911 (NOTE 1)
W1 - FOUR PAIR BUILDING WIRING*

NOTE 1: PEC 62520 INCLUDES Al, A3, AND T1

* FURNISHED BY INSTALLER

Figure 2-40. STARLAN NETWORK Connection to System 25 (With ATL-Type Telephone)

Station Hunting

Description

This feature provides linear, circular, or combinational hunting sequences for calls to busy single-line voice terminals and data terminals.

Calls to a busy terminal may hunt to (only) one other terminal; however, up to five terminals may hunt to the same terminal.

Although hunting is not available to or from multiline terminals, single-line terminals may have their calls covered by multiline terminals.

Station Hunting takes precedence over Coverage. Calls to a single-line voice terminal that is assigned both Station Hunting and Coverage will first hunt. If no hunted-to station is available, the call goes to coverage.

The following are examples of the three types of hunting allowed:

- **Linear Hunting Example:**

  Terminals x, y, and z are arranged for linear hunting as follows: (1) Terminal x hunts to Terminal y, (2) Terminal y hunts to Terminal z, and (3) Terminal z does not hunt.

  An incoming call to a busy terminal in the chain will hunt in one direction only. Hunting will be toward the terminal that does not hunt.

- Circular Hunting Example:

  Terminals x, y, and z are arranged for circular hunting as follows: (1) Terminal x hunts to Terminal y, (2) Terminal y hunts to Terminal z, and (3) Terminal z hunts to Terminal x.

  An incoming call to a busy terminal in the chain hunts in one direction until it finds an idle terminal and then rings at that terminal. Any coverage options assigned to that terminal will then be invoked. If the hunt finds all terminals busy, it will stop at the called terminal. Any coverage options assigned to the called terminal will then be invoked.

- **Combinational Hunting Example:**

  Terminals w, x, and y all hunt to Terminal z.

  An incoming call to a busy w, x, or y Terminal will ring at Terminal z, and any coverage options assigned Terminal z will be invoked. If Terminal z is busy, the call remains at the called terminal. Any coverage options assigned the called terminal will then be invoked.

Considerations

Station Hunting provides several flexible alternatives to ensure that calls do not go unanswered. Note that only calls to *busy* terminals will hunt; once a call begins ringing at a terminal it will remain there unless picked up or covered.

Interactions

The following features interact with Station Hunting.

Attendant Camp-On: When the attendant extends a call to a busy terminal in a hunt group, the call hunts for an idle terminal. If none is found, the call Camps-On to the called terminal.

Callback Queuing: If all stations of a hunt group are busy, the call queues only for the dialed station in the group.

Call Waiting: If all members of a hunt group are busy and the originally-dialed station has Call Waiting, the caller hears special ringback until the station becomes available to answer the call.

Coverage: Station Hunting initially overrides all coverage options. When a call to a voice terminal that has Coverage exhausts the terminal hunting possibilities, coverage is invoked.

Following/Forwarding: Calls signed in at, or forwarded to, a station in a hunt group will hunt and ring an idle station if the home station is busy. If all members of the group are busy and the away, or forwarded-to, station has Call Waiting, the caller hears special ringback until the away station becomes available to answer the call.

A call to a forwarding station in a hunt group will first ring at the away, or forwarded-to, station. After an administered number of rings, the call returns to the hunt group; if all members are busy, the call continues to ring at the away station until a hunt group member becomes available.

Administration Requirements

Voice or Data Terminal Port:

- Assign PDC/DDC of terminal to hunt to next.

Station Message Detail Recording (SMDR)

Description

This feature provides detailed call information records on all incoming and outgoing trunk calls and sends this information to an (optional) output device. Data on inside calls is not collected.

The call records can be used to compute costs, allocate charges, and analyze calling patterns. The output device can be any serial RS-232 compatible DTE device capable of receiving the data (must supply DTR on pin 20) and either printing the call records or storing and analyzing them. (80 character ASCII records are sent to the output device.)

The SMDR RS-232 port interface is provided by a DUART driver (68681). It is a one-way port transmitting data to the output device. No characters are read by the port interface, and no flow control mechanisms are provided. The standard data transmit rate is 1200 bps. (Also operates at 300 bps.)

Call Records

The call records provide detailed information concerning both incoming and outgoing calls. Call detail records are generated during call processing and are sent to the SMDR output device in ASCII format. SMDR records are provided for:

- ***Voice Records:*** The system prints call records for incoming calls and for outgoing calls that exceed a specified duration. For special types of calls such as conference or transferred calls, one call record is reported for each trunk seized, regardless of the number of parties connected to the call. The call's duration is from the time the last digit was dialed until the last person hangs up. No indication is provided that trunks have been bridged together.

- ***Data Records:*** The system prints call records for incoming and outgoing (external) data calls. Calls are considered data calls if they involve a data extension.

The following list describes the SMDR data collected for each call and the number of characters in each field. All information is right justified in its field, unless otherwise indicated. The record is provided in a standard 80-column format. The headings for each record item are noted in bold type. These headings are printed across the top of each page. Page advance is determined by counting lines based on a fixed page length. Each record is followed by a carriage return and a line feed.

The system can provide for the storage of up to 100 SMDR records. If more than 100 records are received while the printer is disconnected, a message "Calls Lost Due To Call Record Overflow" is provided when a printer is re-attached.

The SMDR call detail (Figure 2-41) contains the following information for each call record:

- **TYPE** (Column 1)

  All voice calls are labeled C, data calls are labeled D. (“TYPE” is not printed as a column heading)

- Blank (Column 2)

- **DATE** (Columns 3-10)

  The date the call is originated.

- Blank (Column 11)

- **TIME** (Columns 12-16)

  The time the trunk is seized is listed using a 24-hour clock. For example, 2:01 PM is listed as 14:01. Seconds are truncated.

- Blank (Column 17)

- **CALLED NUMBER** (Columns 18-35)

  For outgoing calls, up to 15 digits may be recorded, excluding the ARS or facility access code but including the 0 or 1 prefix (to identify local and toll calls) and 950-10xx and 10xxx interconnect access codes. Space is allotted for three dashes, one between the fourth and fifth digits from the right, one between the seventh and eight digits from the right, and the other between the tenth and eleventh digits from the right. Numbers longer than 15 digits will be truncated. For Repertory Dialing and Speed Dialing numbers, the facility will be extracted from the stored number and reported under the FAC heading; the number remaining after the facility is extracted will be reported as the called number.

  For a Remote Access call through the System 25 to an outside number, this field contains the outside number dialed by the remote caller.

  An incoming call is identified by the word IN.

- Error Character - Question Mark or Blank (Column 36). Indicates number dialed exceeded 15 digits.

- Blank (Column 37)

- **DUR** (Columns 38-45- Duration)

  For incoming calls, this provides the time between trunk seizure and disconnect, rounded to the nearest second. For outgoing calls, it provides the time between the last digit dialed until the last station on the call hangs up, less an estimated time for call setup (15 seconds), rounded to the nearest second. A call transferred between a number of voice terminals will reflect the total call duration. The maximum time that can be reported is 95 hours, 59 minutes, and 59 seconds.

- Blank (Column 46)

- **FAC** (Columns 47-51 - Facility)

  Indicates the facility used to place the call. For outgoing calls including Speed Dialing numbers, the pooled facility selected by ARS or the facility access code that was dialed (or that corresponds to the facility button that was pressed) is identified. For Remote Access calls through the System 25 to outside numbers, this field contains the number of the outgoing trunk or trunk pool.

  For incoming calls, Personal Line calls, and Remote Access calls to inside numbers, the trunk number is identified.

  If a virtual facility was used to complete the call, the applicable Virtual Facility Code (#190-#199) is identified in this field.

- Blank (Column 52)

- **STN** (Column 53-56)

  Identifies the voice or data terminal responsible for the call. If an account code is entered, the voice terminal where the code is entered is reported. If no account code is entered, the terminal originating an outgoing call is identified, or the last terminal connected to an incoming call is identified.

  For an incoming call to a DGC group that is connected to an announcement but is never answered, 0 will be recorded in the STN field. If the call is answered by a station after receiving announcement, the station answering the call will be recorded.

  If an outgoing call is originated by a tandem tie trunk, the tandem trunk's Facility Access Code (FAC) is recorded in this field. If no FAC exists for this trunk, then the 4-digit trunk number (9xxx) will appear.

  If an outgoing call is originated by a DID trunk, the DID's 4-digit number is recorded in this field.

  For a Remote Access call to an inside number, the called party's number is shown; if the call timed out to the attendant, the attendant's number is shown. If the remote caller calls an outside number, the number of the incoming trunk is presented in this field.

- Blank (Columns 57 ,58)

- **ACCOUNT** (Columns 59-73)

  Lists the Account Code associated with the call, if one was entered. On conference and transferred calls, the first account code entered is recorded and subsequent account code entries are ignored.

  For Remote Access calls to outside numbers, the barrier code is shown.

- Blank (Column 74)
- **PDC** (Columns 75-78)

  Identifies the user responsible for outgoing calls. The user is identified by the call accountability login (##PDC) entered at the originating voice terminal. *If no call accountability is entered, the PDC field is blank.*

Figure 2-42 and Figure 2-43 summarize the Call Record and Call Record Header formats.

Considerations

SMDR provides detailed call information on incoming and outgoing calls. This information can be used to facilitate cost allocation, traffic analysis and detection of unauthorized calls.

Interactions

The following features interact with Station Message Detail Recording.

Account Code Entry: Allows users to have an account code or project number associated with each call record.

Bridging of System Access Button: When an outside call is answered or originated at a Bridged Access (BA) button, the SMDR record for this call will report the bridging station's PDC number under the STN column and the principal station's PDC number under the PDC column. If the Call Accountability feature is used when originating a call at a BA button, the PDC column will contain the *accountable* (entered) PDC number in place of the principal's PDC.

If two bridged stations attempt to originate a call at the same time, and if the call is completed, the PDC number of the station that dialed the first digit is placed in the SMDR records under the STN column.

Direct Group Calling (DGC): For an incoming call to a DGC group that is connected to an announcement and never answered, "0" will be reported in the "STN" field of the call record. If the call is answered by a station after receiving the announcement, that station will be listed in the "STN" field.

Forwarding: When a call is successfully forwarded to an outside number, the call record will contain the forwarding station and forwarded-to station numbers.

When a Remote Access caller activates Forwarding at a System 25 station, the SMDR call record shows the incoming trunk number in the FAC field, the PDC of the forwarding station in the STN field, and the barrier code number in the ACCOUNT field.

Modem Pooling: SMDR records do not reflect modem pool resource usage.

Remote Access: Remote Access calls are fully covered in the SMDR call records.

Tandem Trunking: If an outgoing call is originated by a tandem tie trunk, the tandem trunk's FAC is recorded in the STN field. If no FAC exists for this trunk, then the 4-digit trunk number (9xxx) will appear. No other SMDR fields are affected.

Administration Requirements

System:

- Send SMDR records To SMDR Port (yes or no; default = yes).
- Minimum length (seconds) of calls that are reported by SMDR (10-255; default = 40).

Hardware Requirements

An AT&T Model 475 printer or any standard RS-232 serial 80-column ASCII printer is required for printing the SMDR output. The printer must be dedicated to SMDR to ensure that all calls are recorded. An AT&T Call Accounting System may also be used as the SMDR output device (see below).

The printer can be directly connected to Port 2 of the ZTN129 CPU/MEM (Call Processing Unit/Memory) circuit pack or switched access (either on- or off-premises) can be provided. Connection is the same as described for the SAT.

SMDR port parameters are as follows:

- No parity; bit is set to zero.
- 1 start bit, 1 stop bit, and 7 data bits.
- Baud rate defaults to 1200 (can be set to 300).
- DTR (data terminal ready) required from printer.
- RTS (ready to send) and CTS (clear to send) not required.
- No flow control.

Detailed connection information is provided in Figures 2-44 through 2-48.

Maximum cabling distances are provided in Section 5, "Technical Specifications."

	DATE	TIME	CALLED NUMBER	DUR	FAC	STN	ACCOUNT	PDC
C	10/08/85	11:14	1-232-566-1321	00:15:41	9	1794	123489764321341	4271
C	10/08/85	11:14	IN	00:09:05	2145	1324		
C	10/08/85	11:15	1-322-564-1376	00:29:50	9	1744	766544	3254
D	10/08/85	11:17	1-222-564-2171	00:10:45	9	2001		
C	10/08/85	11:19	IN	00:05:32	3214	1744		
C	10/08/85	11:20	IN	00:29:45	2342	3455		
C	10/08/85	11:34	-242-563-1324	00:19:00	9	1677	76322	3422
D	10/08/85	11:44	555-4541	00:05:35	9	2312	323489764321341	3422
D	10/08/85	11:50	-252-514-3176	00:19:45	9	3455		
C	10/08/85	12:14	IN	00:25:42	2145	1492		
C	10/08/85	12:24	-222-566-2544	00:10:35	9	1244	123489764321341	4271
C	10/08/85	13:27	-333-513-1376	00:15:05	9	3566		
C	10/08/85	13:14	1-244-564-3121	00:09:40	9	2001		
C	10/08/85	13:23	IN	00:15:45	3414	1566		
C	10/08/85	13:28	IN	00:19:35	3421	3421		
C	10/08/85	13:33	1-222-516-1176	00:19:40	9	1492	123489764321341	4271
C	10/08/85	13:45	555-2541	00:09:05	9	3655		
C	10/08/85	14:14	-222-563-4321	00:20:42	9	4321	763444	3465
D	10/08/85	14:24	-343-516-2574	00:10:05	9	1244		
C	10/08/85	14:34	555-3141	00:09:45	9	4633		
C	10/08/85	14:43	IN	00:19:32	3214	2351		
C	10/08/85	15:14	-343-564-1321	00:20:45	9	1794	123489764321341	4271
C	10/08/85	15:16	-222-566-1321	00:19:45	9	1794		
C	10/08/85	15:19	IN	00:19:45	2342	1794		

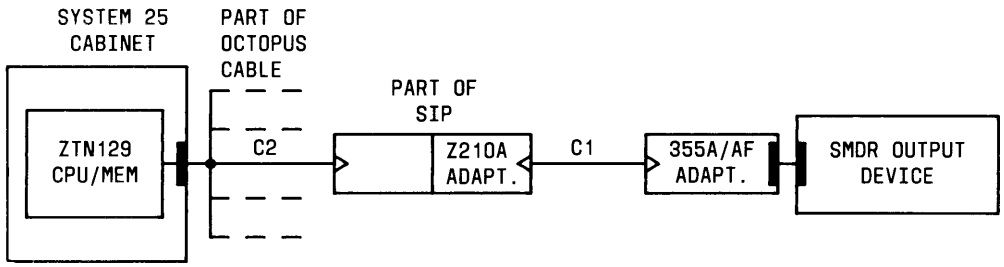
Figure 2-41. Typical SMDR Call Detail Report

ASCII CHARACTER POSITION (Column Number)	DESCRIPTION	VALID CHARACTERS
01	Call Type	(3 or D
02	Space	
(-)3-04	Date: Month	0-9
05	/	/
06-07	Date:Day	0-9
08	/	/
09”10	Date:Year	0-9
11	Space	
12-13	Time: Hour	0-9
14		
15-16	Time:Minute	0-9
17	Space	
18-35	Dialed Number	0-9, Space, -, IN
36	Error Character	?, or Space
37	Space	
38-39	Duration:Hour	0-9
40		
41-42	Duration: Minute	0:9
43		
44-45	Duration: Second	0-9
46	Space	
47-51	Facility	Space, 0-9, #
52	Space	
53-56	Station	0-9, Space
57-58	Space	
59-73	Account Code	0-9, Space, #, ?
74	Space	
75-78	Personal Dial Code	0-9, Space
79		Carriage Return
80		Line Feed

Figure 2-42. SMDR Call Record Format

ASCII CHARACTER POSITION	DESCRIPTION
00	(top of form)
01-03	Space
04-07	DATE
08-12	Space
13-16	TIME
17-23	Space
24-29	CALLED
30	Space
31-36	NUMBER
37-38	Space
39-41	DUR
42-48	Space
49-51	FAC
52-54	Space
55-57	STN
58-62	Space
63-69	ACCOUNT
70-76	Space
77-79	PDC
80	(Carriage Return)
81	(Line Feed)

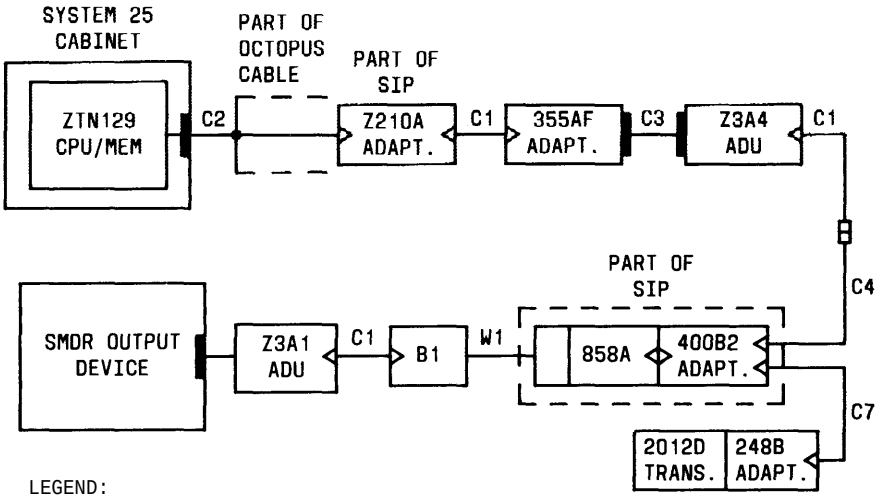
Figure 2-43. SMDR Call Record Header Format



LEGEND:

C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
355A ADAPTER - RS-232 PLUG TO MODULAR JACK - PEC 2750-A24
355AF ADAPTER – RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-44. SMDR Output Equipment—On-Premises Direct Connections (Sharing Same AC Outlet)



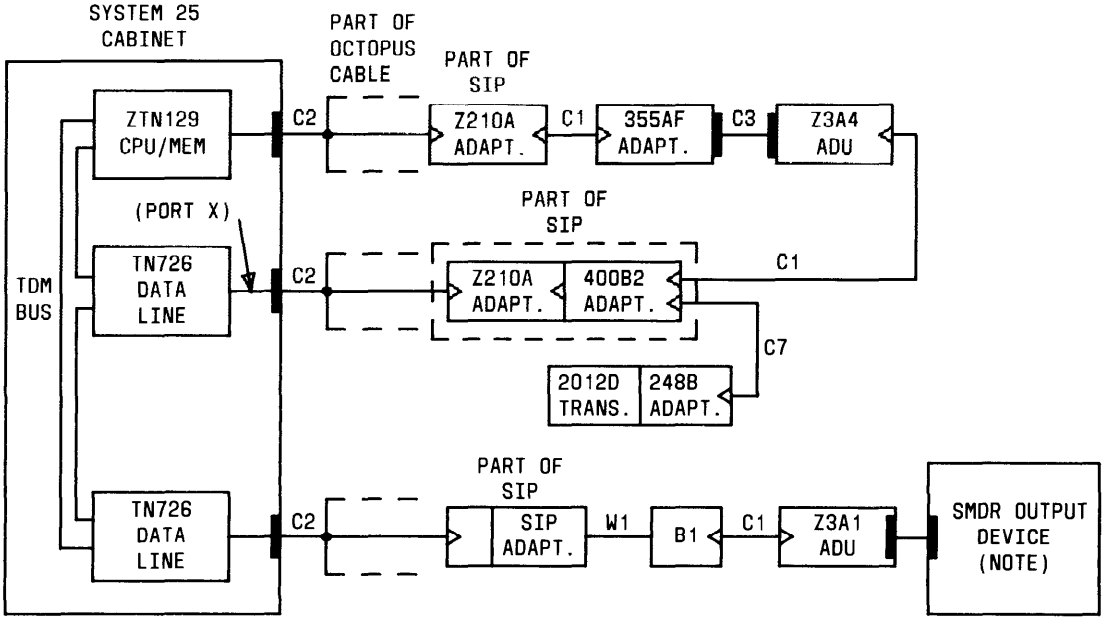
LEGEND:

B1 - TYPICAL-103A CONNECTION BLOCK*
C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
W1 - 4-PAIR INSIDE WIRING CABLE*
Z3A1 ADU - EQUIPPED WITH 3-FOOT PLUG-ENDED EIA CORD – PEC 2169-001
C3 - EIA CROSSOVER CABLE (M7U-87)
C4 - ADU CROSSOVER CABLE (D8AM-87)
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD
C1 - MODULAR CORDS (2) (D8W-87)
C7 - MODULAR POWER CORD (D6AP-87)
2486 ADAPTER - MODULARIZES 2012D TRANSFORMER
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE

PEC 62515

* - FURNISHED BY INSTALLER

Figure 2-45. SMDR Output Equipment—On-Premises Direct Connections (Greater Than 50 Feet From System Cabinet or Nonsharing Same AC Outlet)



LEGEND:

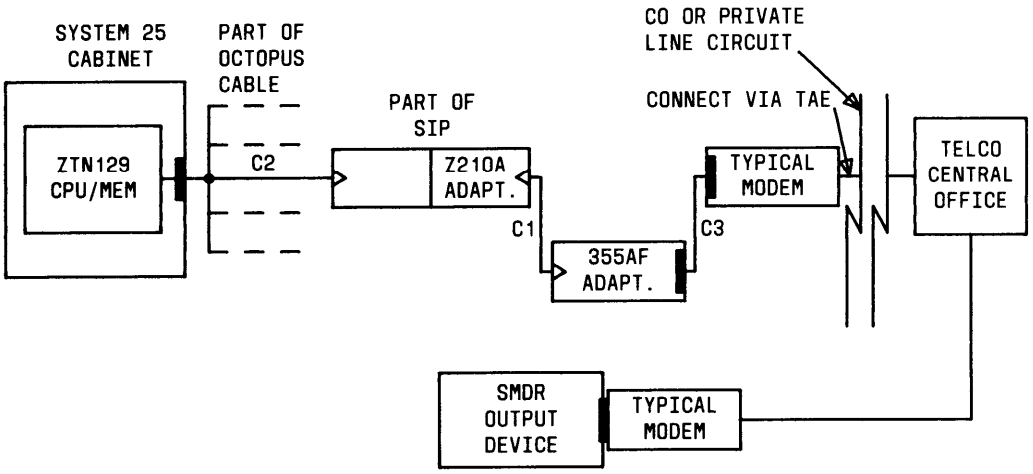
B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
W1 - 4-PAIR INSIDE WIRING CABLE*
Z3A1 ADU - EQUIPPED WITH 3-FOOT PLUG-ENDED EIA CORD – PEC 2169-001
C3 - EIA CROSSOVER CABLE (M7U-87)
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD
C1 - MODULAR CORDS (2) (D8W-87)
C7 - MODULAR POWER CORD (D6AP-87)
248B ADAPTER - MODULARIZES 2012D TRANSFORMER
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE

PEC 62515

* - FURNISHED BY INSTALLER

NOTE: OUTPUT DEVICE OR MULTILINE VOICE TERMINAL WITH DATA BUTTON DIALS PORT X DDC TO ESTABLISH DATA CONNECTION TO ZTN-82.

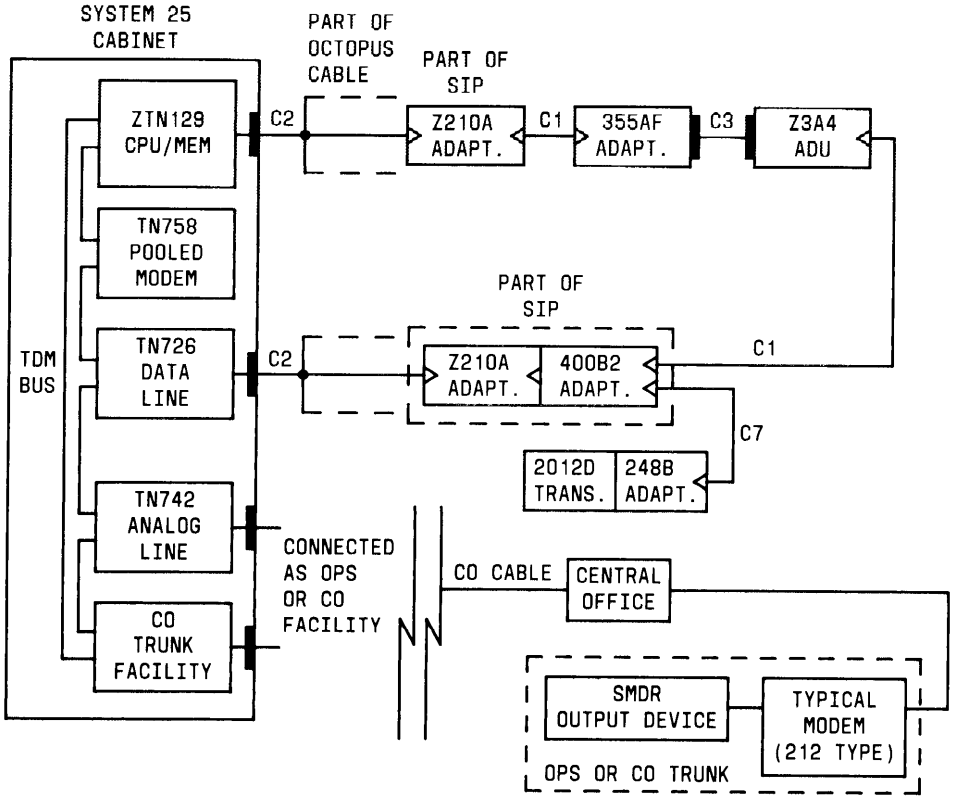
Figure 2-46. SMDR Output Equipment—On-Premises Switched Connections



LEGEND:

C1 –MODULAR CORD (D8W-87) - PEC 2725-07G
C2 –OCTOPUS CABLE (WP90780) - PEC 2720-05P
C3 –EIA CROSSOVER CORD (M7U-87) - PEC 2724-30C
355AF ADAPTER – RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-47. SMDR Output Equipment—Off-Premises Direct Connections



LEGEND:

C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C3 - EIA CROSSOVER CABLE (M7U-87) - PEC 2724-30G
OPS - OFF-PREMISES STATION
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD - PEC 2169-004
248B ADAPTER - MODULARIZES 2012D TRANSFORMER
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE
C7 - MODULAR POWER CORD (D6AP-87)
PEC 21691

Figure 2-48. SMDR Output Equipment—Off-Premises Switched Connections

Station-to-Station Message Waiting

Description

This feature allows pairs of multiline voice terminal users to signal each other with Message Waiting (MSG WAIT) buttons and associated green status LEDs at each terminal. When either user presses the MSG WAIT button, the LEDs light at both stations. This arrangement enables one user to inform the other user that a message is waiting; it can also be adapted to other two-way signaling purposes such as “come to my off ice.”

After the MSG WAIT LEDs have been turned on, they can be turned off by operation of the button at either terminal. The receiver of a station-to-station message should normally turn off the LEDs as soon as the message is understood so that the link is restored to an idle condition and is ready for use again.

No talking path or audible alerting is associated with this feature.

Considerations

This feature is functionally similar to, but separate from, the Coverage Message Waiting and Attendant Message Waiting features.

A station can be a member of more than one Station-to-Station Message Waiting pair, but must have a separate button for each pair.

This feature is not associated with the built-in MESSAGE (or MSG) indicators of many System 25 voice terminals. It lights only the LEDs of the feature buttons assigned to Station-to-Station Message Waiting.

Administration Requirements

Voice Terminal Port:

- For each pair of stations that are to share this feature, assign a MSG WAIT button with associated status LED at each station.

  Note: MSG WAIT buttons always assigned to *pairs* of stations for use only between the two stations. If station A needs to signal stations B and C, station A must have a separate MSG WAIT button for each.

System Administration

Description

The software that controls System 25 operation consists of tables located in system memory. These tables contain data associated with:

- Trunk, Station, and Auxiliary Equipment Ports
- System Parameters
- Direct Group Calling Groups
- Toll Calls Allowed Lists
- Peripheral Equipment Data Communications Parameters
- Automatic Route Selection.

Collectively, these software tables are referred to as translations. The system comes equipped with default translations data; when *full-default cold started,* the default translations are copied into translation memory.

System Administration is the process of managing the translations by making changes to modify system operation to meet customer requirements.

The *System* 25 *Implernentation Manual* describes how a system can be configured to meet specific customer needs. Information about a desired configuration is recorded on a set of forms that are used when entering the initial system translations (i.e., initializing the system). These forms are filed in the Administration Records Binder and provide the basis for on-going record keeping. Modification of initial assignments can be made to meet changing customer needs.

The system provides an EIA RS-232 interface to a System Administration Terminal (SAT), the primary means of entering and modifying translations.

System 25 administration consists of:

- ***Centralized Administration:*** Configuration of the system and assignment of feature-related parameters, including assignment of feature buttons on voice terminals. Centralized Administration is performed via the SAT.
- ***Advanced Administration:*** The Advanced Administration Software (AAS) package is a major improvement in system management. It provides the System Administrator a user-friendly, powerful tool for accurately and quickly making changes in voice/data terminal assignments, coverage, access codes, and other system functions such as ARS.

  Two sets of software are available for Advanced Administration. One set operates on a PC6300 (with 640K RAM) and provides an alternative to use of the SAT input terminal. The other runs on the Master Controller (UNIX PC) as part of the "Integrated Solution."

A main menu gives the user ready access for these tasks:

— Adding/changing/removing voice station assignments

— Adding/removing users to and from coverage groups

— Saving translations.

A significant advantage of the AAS package is that it can be used either at the same location or can be used remotely via a dial-up connection.

Considerations

The default system administration password **(systemx5)** can be changed through an administration item. Note, however, that a system warm start or cold start will reset the password to this default.

Hardware Requirements

The System Administration Terminal (SAT) is a Model 703 Data Terminal (see Figure 2-49). It is a general purpose asynchronous full duplex printing data terminal with an RS-232 interface for data entry and retrieval. It provides a paper record of all transactions. When located within 50 feet of the system cabinets, it can be directly connected to channel 1 on the ZTN129 CPU/MEM (Call Processing Unit/Memory) CP. Either on-premises or off-premises access to the administration port is supported. The terminal operates at a speed of 1200 bps (1200 baud).

Administration port parameters are as follows:

- No parity; bit is set to zero.
- 1 start bit, 1 stop bit, and 7 data bits.
- Autobaud is invoked when carriage return is pressed (300 or 1200).
- DTR (data terminal ready) required from terminal.
- RTS (ready to send) and CTS (clear to send) not required.
- No flow control.

The Model 703 requires 115V ac 60-hertz commercial power from a 3-wire grounded outlet.

The terminal should be located on a flat surface such as a desk or table top. It is approximately 12 inches wide, 9 inches long, and 3 inches high.

The Model 703 keyboard generates ASCII codes. The terminal produces two audible tones to indicate the completion of activities.

- Short Tone—A tone of less than one half-second indicates the normal termination of an operation
- Long Tone—A one-second tone indicates that an error or an abnormal operating condition has been detected.

The *Model 703 SAT Supplement* contains a complete set of operating instructions for the Model 703 Data Terminal. This document may be of use to customers who want to use the terminal for other purposes in addition to system administration. All the information needed to use the terminal as a SAT is included in the *R2V1 Administration Manual.*

The SAT can be connected to the system cabinets in several different ways:

- A direct connection within 50 feet when sharing the same AC outlet as the system cabinets
- A direct on-premises connection at a distance greater than 50 feet from the system cabinets
- A direct off-premises connection via the Central Office (OPS or CO trunk)
- An on-premises switched connection
- An off-premises switched connection.

Maximum cabling distances from the system cabinets are provided in Section 5, “Technical Specifications.”

Installation details are provided in the *System* 25 *Installation and Maintenance Manual.*

The SAT may also be provided by the customer. It must be a RS-232 compatible terminal that has a 25-pin connector providing signal on DTR (pin 20). In addition, it should have the following characteristics:

Display: The minimum display size is 16 lines by 80 columns. The port provides both carriage return and line feed characters to position the cursor at the start of the next line. Destructive scrolling is also expected (new lines added at the bottom of the screen and top-most lines disappear). Full duplex operation is required. Alphabetic ASCII characters in both upper-case and lower-case will be sent to the SAT, along with ASCII numerals and some basic ASCII symbols. The device used must be capable of displaying ASCII alphabetic characters when either upper-case or lower-case characters are received. However, upper-to-lower case mapping (or vice-versa) for display is acceptable since no meaning is associated with case.

Keyboard: The administration port requires ASCII alphanumeric characters as well as some symbol characters. If the keyboard generates only upper-case or only lower-case alphabetic characters the administration port will respond appropriately, since upper and lower case input is considered identical. The SAT should be capable of sending the following ASCII Characters.

A-Z or a-z
0-9
*,#

?
BACKSPACE
" RETURN

The data transfer rate is set when a carriage return character is received by the administration port. There are two supported transfer rates: 1200 bps and 300 bps.

Refer to the *R2V1 Administration Manual* for administration procedures and additional information.

SAT Connection Information

Detailed connection information is provided in the following figures:

- Figure 2-50—SAT On-Premises Direct Connections (Sharing Same AC Outlet)
- Figure 2-51—SAT On-Premises Direct Connections (Greater Than 50 Feet from System Cabinet or Not Sharing Same AC Outlet)
- Figure 2-52—SAT On-Premises Switched Connections
- Figure 2-53—SAT Off-Premises Direct Connections
- Figure 2-54—SAT Off-Premises Switched Connections

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” in Section 4.

Maximum cabling distances from the system cabinets to the SAT are provided in Section 5, “Technical Specifications.”

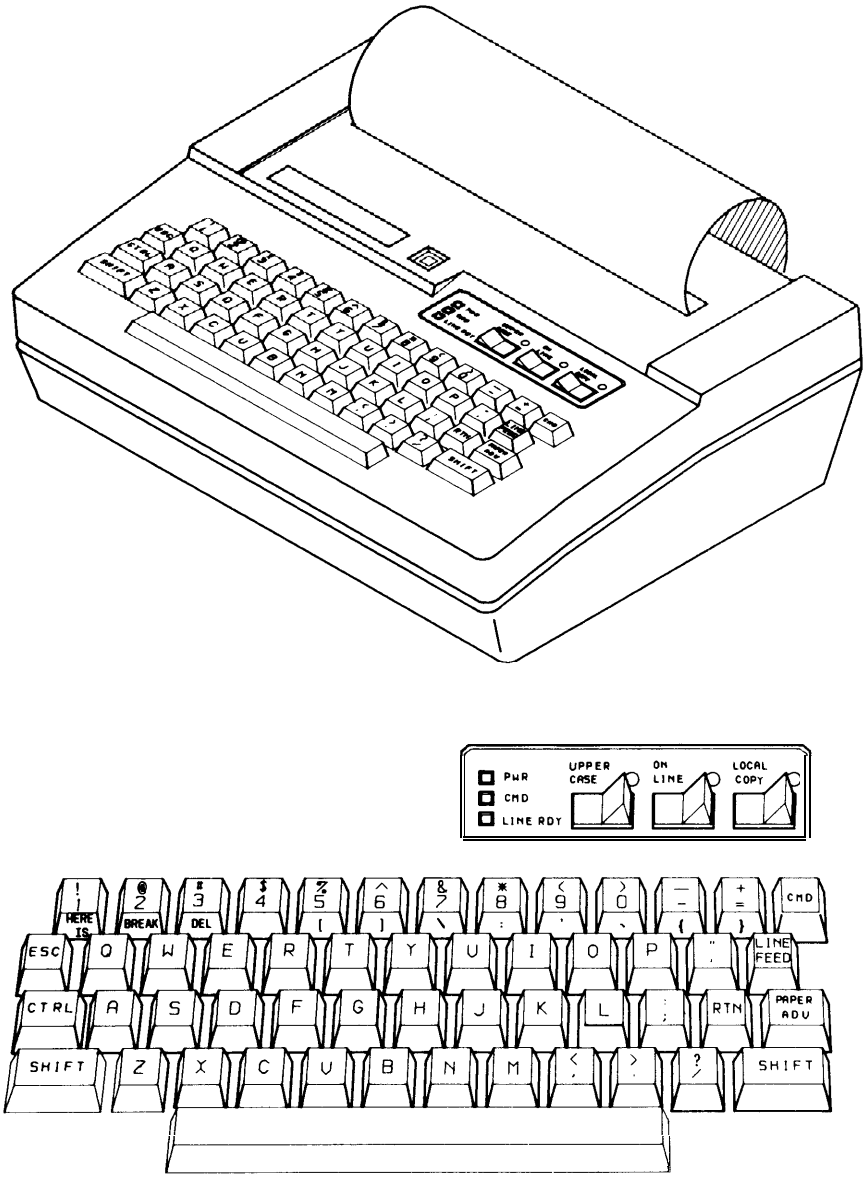
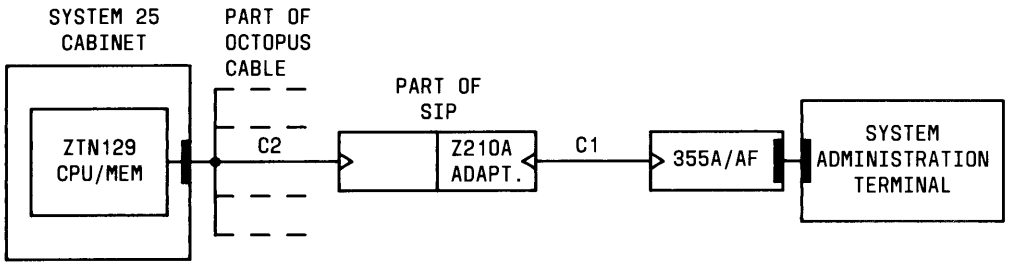


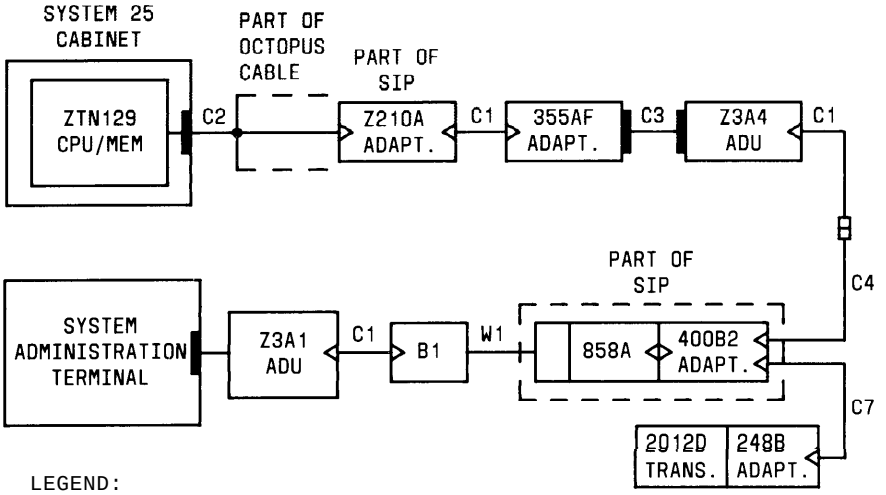
Figure 2-49. Model 703 System Administration Terminal



LEGEND:

C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C 2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
355A ADAPTER - RS-232 PLUG TO MODULAR JACK - PEC 2750-A24
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-50. SAT On-Premises Direct Connections (Sharing Same AC Outlet)



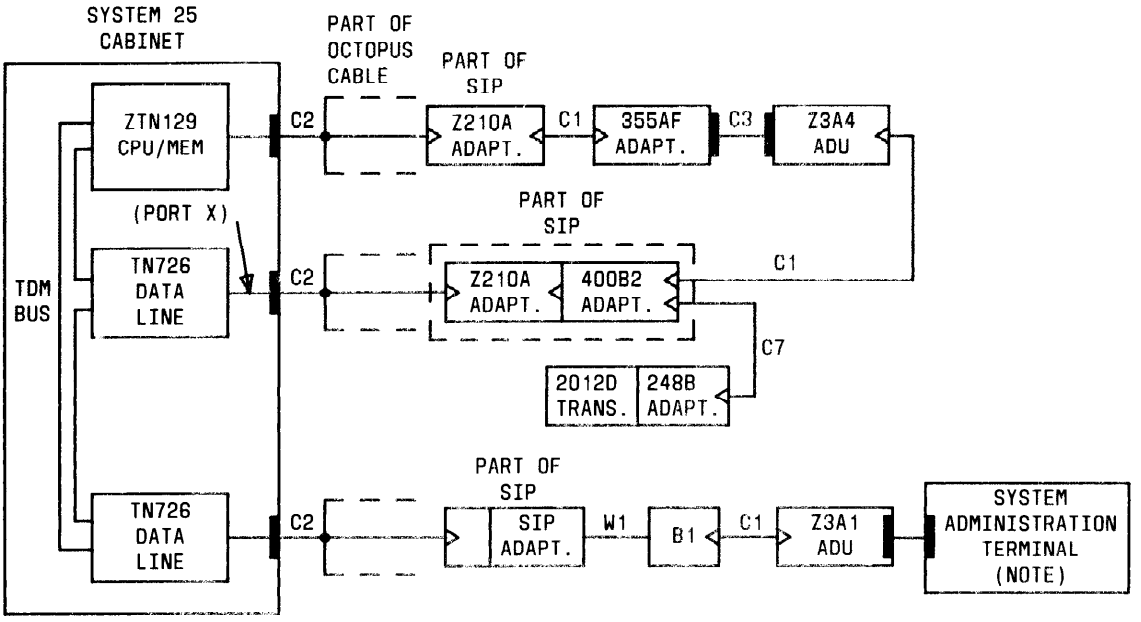
LEGEND:

B1 - TYPICAL-103A CONNECTION BLOCK*
C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
W1 - 4-PAIR INSIDE WIRING CABLE*
Z3A1 ADU - EQUIPPED WITH 3-FOOT PLUG-ENDED EIA CORD -PEC 2169-001
C3 - EIA CROSSOVER CABLE (M7U-87)
C4 - ADU CROSSOVER CABLE (D8AM-87)
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD
C1 - MODULAR CORDS (2) (D8W-87)
C7 - MODULAR POWER CORD (D6AP-87)
248B ADAPTER - MODULARIZES 20120 TRANSFORMER
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE

PEC 62515

* - FURNISHED BY INSTALLER

Figure 2-51. SAT On-Premises Direct Connections (Greater Than 50 Feet From System Cabinet or Nonsharing Same AC Outlet)



LEGEND:

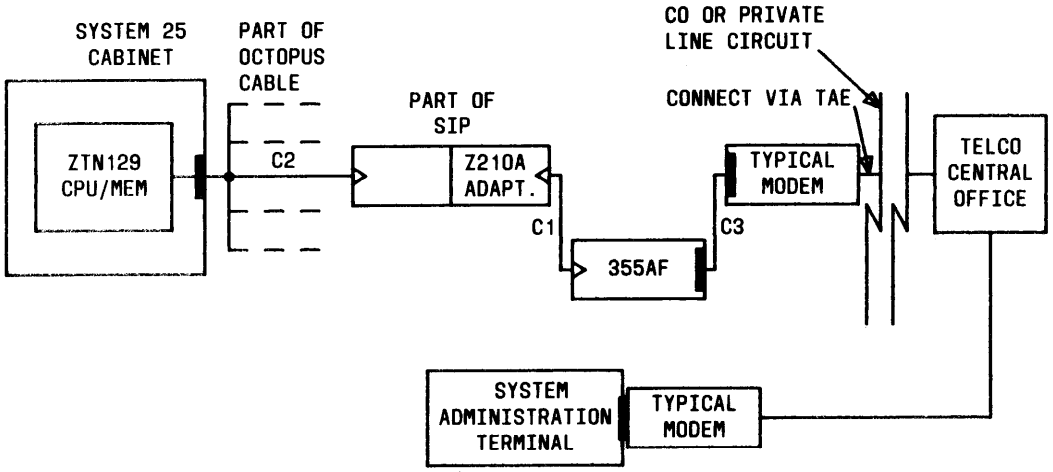
```
              B1 - TYPICAL-103A CONNECTING BLOCK*
              C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
              C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
              W1 - 4-PAIR INSIDE WIRING CABLE*
        Z3A1 ADU - EQUIPPED WITH 3-FOOT PLUG-ENDED EIA
                   CORD - PEC 2169-001
              C3 - EIA CROSSOVER CABLE. (M7U-87)
   355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK
        Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD
              C1 - MODULAR CORDS (2) (D8W-87)
              C7 - MODULAR POWER CORD (D6AP-87)
    248B ADAPTER - MODULARIZES 2012D TRANSFORMER
   400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE
```

PEC 62515

* - FURNISHED BY INSTALLER

NOTE: SYSTEM ADMINISTRATION TERMINAL OR MULTILINE VOICE TERMINAL WITH DATA BUTTON DIALS PORT X DDC TO ESTABLISH DATA CONNECTION TO ZTN-82.

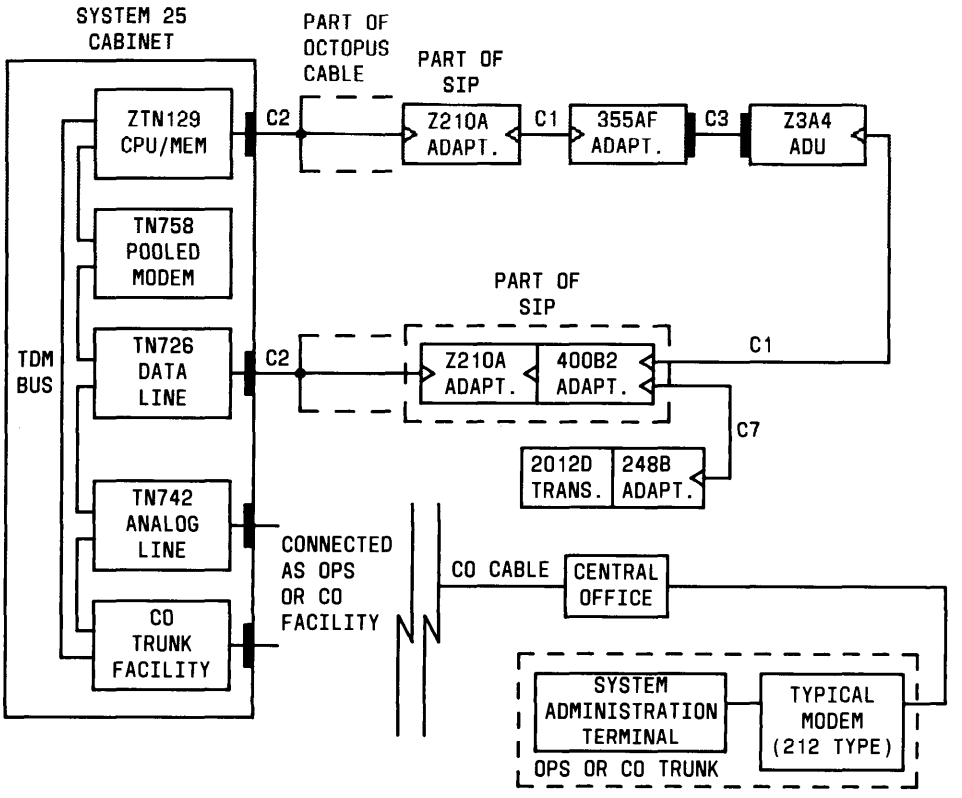
Figure 2-52. SAT On-Premises Switched Connections



LEGEND:

C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C3 - EIA CROSSOVER CORD (M7U-87) - PEC 2724-30C
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-53. SAT Off-Premises Direct Connections



LEGEND:

C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C3 - EIA CROSSOVER CABLE (M7U-87) - PEC 2724-30G
OPS - OFF-PREMISES STATION
355AF ADAPTER - RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25
Z3A4 ADU - EQUIPPED WITH 3-FOOT RECEPTACLE-ENDED EIA CORD - PEC 2169-004

PEC 21691:
2486 ADAPTER - MODULARIZES 2012D TRANSFORMER
40062 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE
C7 - MODULAR POWER CORD (D6AP-87)

Figure 2-54. SAT Off-Premises Switched Connections

System Maintenance

Description

The primary objective of System 25 maintenance is to detect, report, and clear troubles as quickly as possible and with minimum disruption to normal service. This goal is supported by periodic automatic diagnostic tests and fault detection hardware. System design allows most troubles to be resolved to the circuit pack level.

System 25 hardware and software are organized as independent units or maintenance objects. Each maintenance object is normally a separately replaceable unit. These units include circuit packs, power units, fans, voice and data terminals, cross-connect hardware, auxiliary, and peripheral equipment.

There are two general categories of system errors: system-detected errors and user-reported problems. The system can automatically detect and log errors without human intervention. For system-detected errors, an Alarm LED on the Attendant Console is lighted if the error qualifies as a Permanent System Alarm (a serious error). Most alarms can be verified by checking the LEDs located on the front edge of the system circuit packs. (At least one Red LED will be on.) User-reported problems are usually detected at individual voice and data terminals and are often related to alarmed conditions.

Alarms may be retired automatically and can also be cleared manually. After a trouble has been cleared, the system retests the previously faulty area. If the fault is no longer present, the error message (and alarm, if applicable) is cleared. It is not necessary for maintenance personnel to retire alarms after a problem has been fixed. However, they may clear error messages and alarms by entering the proper commands at the System Administration Terminal.

System Errors And Alarms

If a maintenance object fails periodic tests, the system automatically generates an error record that is placed in one of three software tables (error logs). The failure may be classified as a Permanent System Alarm or as an unverified failure that never becomes a Permanent System Alarm. A Permanent System Alarm causes the Alarm LED on the Attendant Console to light. This alarm indication is a signal to the attendant to contact maintenance personnel.

System alarms are classified as:

- ***Permanent System Alarms:*** Failures that cause degradation of service and require immediate attention. These alarms cause the Alarm LED on the Attendant Console to light and an alarm record to be stored in the Permanent System Alarm error log.

- ***Transient System Errors:*** Potential failures that may cause degradation of service. These do not light the Alarm LED on the Attendant Console. These are errors that have not been verified by system self-tests, and are not yet serious enough to be classified as Permanent System Alarms.

If an error that begins as a Transient System Error is verified or reaches a threshold level of severity, it is reclassified as a Permanent System Alarm.

Transient system errors are stored in the Transient System Error log. The system can store a combined total of 40 Permanent System Alarms and Transient System Errors in the error tables.

- ***Most Recent System Errors:*** The ten Most Recent System Errors are recorded by the system, regardless of their level of severity. These are stored in the Most Recent System Errors log.

Error Logs

The three error logs can be displayed via the System Administration Terminal. The data in the log is useful in diagnosing and analyzing troubles, particularly when the problem has not yet caused an alarm or when alarms cannot be retired by replacement of maintenance objects.

The error log is historical in nature. It lists faults that have not been resolved as well as past alarms, and provides a profile of system maintenance.

Automatic Maintenance Tests

There are two kinds of maintenance testing initiated (only) by the system:

- Periodic
- Demand

Periodic tests are run by the system at fixed intervals. The tests do not affect service.

Demand tests are run by the system when it detects a condition requiring a need for testing. Demand tests are only performed when errors are detected. Maintenance personnel cannot initiate these tests.

For additional information, see *AT&T System 25 Installation and Maintenance Manual.*

Tandem Trunking

Description

Tandem trunking provides an enhanced networking capability for System 25. With this feature, tie trunks can be used to call through System 25 to reach another switching system (CO or PBX). Calls may be completed over on-network or off-network facilities.

To be treated as tandem trunks, tie trunks must be assigned trunk numbers beginning with the digit 9. Incoming calls on these trunks may route out of System 25 over ground start, loop start, or tie trunks. Tandem trunks can gain access to outgoing facilities either indirectly (by the ARS feature) or directly (by dial access).

System 25's Tandem Trunking feature does *not* support traveling class marks or centralized attendant service. Users cannot activate most System 25 features or services at either the tandem or far-end terminating switch.

Considerations

The use of tandem trunking with tie trunks provides a cost-effective alternative to toll calling between branches.

Interactions

The following features interact with Tandem Trunking.

Automatic Route Selection: Tandem trunk calls that route outbound via ARS receive the same treatment as calls originated by a System 25 station, with one exception. If all facilities in a routing pattern are busy, call queuing is *not* provided. In this case, busy tone is returned to the calling party.

The second digit of the trunk number is used to specify the trunk's "station" Facility Restriction Level (FRL) for use with ARS. FRLs may be specified as follows:

Trunk Number Range	Second Digit	Station FRL
9000-9199	0 or 1	0
9200-9399	2 or 3	1
9400-9599	4 or 5	2
9600-9999	6, 7, 8, or 9	3

To gain access to an ARS routing facility, the tandem trunk's "station" FRL must be equal to or greater than the route's FRL. Thus, a tandem trunk with an FRL of 0 has the least ARS privileges, while an FRL of 3 provides the most privileges. If the restriction level of the tandem trunk is less than all route FRLs, reorder tone is returned to the calling party.

Dial Access: No toll restriction is provided for tandem trunk calls. However, access to outgoing facilities can be controlled via the "allow dial access" option in the *outgoing* trunk's administration. Tandem Trunk calls receive the following treatment when attempting dial access of System 25 facilities:

- If the requested trunk pool is dial accessible, an outgoing trunk is selected and the call proceeds normally.
- If the requested trunk pool s not dial accessible, reorder tone is returned to the calling party.

Any attempt to dial an outgoing trunk pool by non-tandem tie trunks (that is, tie trunks whose trunk number does not begin with 9) results in reorder tone being returned to the calling party.

Paging System Access: Tandem trunks can access paging ports (auxiliary or CO trunk interface), as long as the paging ports are dial-accessible.

Station Message Detail Recording (SMDR): After accessing an outgoing facility, the tandem trunk's Facility Access Code (FAC) will be recorded in the STN field on the call's SMDR record. If the tandem trunk has no FAC, then the 4-digit trunk number (9xxx) will be recorded in the STN field. All other SMDR fields are unaffected.

Administration Requirements

Except for the need to specify the trunk number as described above, administration of tandem trunks is the same as for any other tie trunk.

Hardware Requirements

Only dial-in tie trunks (types 1003-1008) may be used for tandem trunking.

Test

Description

This feature provides users of the 7300H series voice terminals the ability to test their terminals.

Placing the Test/Program (T/P) switch in the “T” position causes all red and green LEDs to light alternately and the tone ringer to sound. If the terminal has a display module, the following responses also occur:

- The display shows 16 darkened squares.
- After the Test switch is returned to its normal position, the Local Display alarm clock produces 3 short beeps.

Considerations

The Test feature assures users that all visible indicators (LEDs and display screen) and audible signaling devices (tone ringer, built-in speaker, and timer alarm) of their terminals are working properly.

The Test switch on some voice terminals is spring loaded; upon release, the switch returns to a normal on-line position. On other terminals, the switch must be manually returned to the center (normal on-line) position.

Third-Party Call Setup

Description

The Third-Party Call Setup feature allows a data terminal (the *third party)* to set up, via Command Mode, a call between an on-premises voice or data terminal (the source) and another voice or data terminal (the *destination;* can be on- or off-premises). Once the call has been set up, the *third-party* drops off and is *not* included in the call.

Each *third-party* data terminal may be administered to have a particular *source* terminal "associated" with it. This association allows an abbreviated form of dialing when activating the Third-Party Call Setup feature. Through further administration, the *third-party* data terminal may be given permission to set up calls for *any* source terminal, for only the *associated* source terminal, or for *no* source terminal (feature disabled).

When the user successfully activates Third-Party Call Setup and has dialed all digits correctly, the following occurs, depending on the *source* terminal type:

- **Voice Terminal (source)**

  The *source* voice terminal (if not busy) receives priority ringing. A priority ringing cycle consists of two short bursts followed by one long burst. The *source* terminal's handset must be picked up within three ringing cycles; the *destination* terminal will then be called. Regardless of the call outcome, the *third-party* data terminal displays the message **CONFIRMED** and **DISCONNECTED** immediately after calling the *destination* terminal.

  If the *source* terminal's handset is not picked up within three priority ringing cycles, the third-party data terminal displays the messages **NO ANSWER** and **DISCONNECTED.** If the *source* phone is busy, the *third-party* data terminal displays **BUSY** and **DISCONNECTED.**

  A call to a *source* terminal that has the Hands-Free Answer on Intercom (HFAI) feature activated results in the automatic answering of the *source* end, and the *destination* terminal will be called. The *third-party* data terminal displays **CONFIRMED** and **DISCONNECTED.**

- **Data Terminal (source)**

  If the *source* terminal is a data terminal whose speed is set to the highest optioned speed of the data port, the message **REMOTE SETUP** is displayed at the *source* terminal and the **CONFIRMED/DISCONNECTED** message is displayed at the *third-party* data terminal. The *destination* terminal will then be called automatically from the *source* terminal. If the *source* terminal and *destination* terminal are compatible, a data connection is established.

Since the System 25 does not provide call progress tone detection for an off-premises call (can't detect second dial tone, for example), pause characters should be inserted at appropriate places in the dialed digit string. In addition, Third-Party Call Setup calls are subject to the administered restrictions assigned to the *source* voice or data port. For

example, if the *source* terminal is restricted to on-premises calls only, a call to an off-premises *destination* terminal will be blocked.

Setting Up A Third-Party Call

To set up a call from the *third-party* data terminal, the user selects **<Voice call>** from the entry-level Command Mode menu (see Figure 2-51). The user then enters the characters as required to call the *destination* terminal. Calls may be completed as follows.

Note: If a character is entered incorrectly, the ASCII character backspace (BS or CTRL-H keys) or underscore (_) may be used to cancel a previously entered character.

Calling a Destination Terminal (source terminal is NOT associated with the third-party data terminal)

After the user has selected **<Voice call>** from the Command Mode menu, a DIAL: prompt is displayed on the *third-party* data terminal. The user has 15 seconds to begin entering the digits to be dialed before being disconnected. The format of the digits following the DIAL: prompt is shown below:

DIAL: {Destination }F{Source}

The Destination number must include all digits required to call the *destination* terminal and may contain facility access codes, Speed Dialing codes, and pauses (*). An “F” may be entered immediately following the Destination digits; this character is used to separate the *Destination* number from the *Source* number. The Source number must be a Personal Dial Code when the *source* is a voice terminal, or a Data Dial Code when the *source* is a data terminal. Floating PDCs are not allowed.

If the user enters the Destination number but not the Source number, the system prompts as indicated below:

FROM: {Source}

The user must then enter the Source number.

Calling a Destination Terminal (source terminal is associated with the third-party data terminal)

Following the DIAL: prompt the user enters the Destination number only. (System 25 will automatically select the associated PDC or DDC as the Source number.)

If the *third-party* data terminal is permitted to establish calls for any *source* terminal, the format {Destination} F{ Source} must be used to set up calls for any terminal except the associated *source* terminal.

Considerations

With this feature, computer-based telemarketing or other calling applications can set up calls for the user.

Interactions

The following features interact with Third-Party Call Setup.

Account Code Entry, Forced (FACE): If the source station is FACE-restricted, the third-party data terminal must prefix the outside destination number with *0 and an account code.

Coverage: Third-Party Call Setup calls to the *source* terminal will not be directed to a coverage station. If the *source* terminal is not answered before coverage is invoked, the call is dropped and the **NO ANSWER/DISCONNECTED** messages are displayed at the *third-party* data terminal.

Expert Mode: Refer to the Expert Mode feature description for additional methods in dialing when using the Third-Party Call Setup feature.

Following/Forwarding: If calls to a *source* terminal have been redirected to another terminal (via Following or Forwarding), Third-Party Call Setup calls will be redirected, Note, however, that the maximum of three ringing cycles (combined cycles at the home and away terminals) still applies before the system drops the call and displays the **NO ANSWER/DISCONNECTED** messages at the *third-party* data terminal.

Station Message Detailed Recording (SMDR): SMDR records will be generated for Third-Party Call Setup calls just as if they were placed by the *source* terminal.

Administration Requirements

- Data Port:

  The user's data terminal may be administered to have a particular *source* terminal associated with it. This allows the abbreviated form of dialing when activating the Third-Party Call Setup feature.

  A *source* terminal may be associated via administration with several *third-party* data terminals. A *third-party* data terminal, however, may be associated with only one *source* terminal (voice or data).

  In addition, the *third-party* data terminal can be administered so that calls may be established for:

  — Any voice or data *source* terminal.

  — One associated *source* terminal only.

— No *source* terminals (feature disabled; default).

- AT&T STARLAN NETWORK Access:

  Ports administered on the STARLAN INTERFACE circuit pack *may not* have a particular *source* terminal associated with them. Depending on the administration parameters enabled, *third-party* data terminals on the STARLAN INTERFACE circuit pack may establish calls for:

  — Any voice or data *source* terminal

  — No *source* terminals (feature disabled; default).

Tie Trunks

Description

Tie trunks provide private communications links between System 25 and other PBXs.

Incoming tie trunk calls may be directed to the attendant, to a voice terminal, or to a data endpoint. Service may be either automatic, immediate dial, delay dial, or wink start. Dial pulse or touch-tone signaling is supported on both incoming and outgoing calls (and may be different for incoming and outgoing calls).

Considerations

Tie Trunks provide for efficient communications between company employees at different locations. This provides a private network whose control and utilization can be managed.

Tie trunks can be administered for tandem trunking. This arrangement enables users to call through an intermediate System 25 to a remote System 25 or other PBX. Refer to the "Tandem Trunking" feature description for more information.

Interactions

The following features interact with Tie Trunks.

Automatic Route Selection (ARS): Immediate dial tie trunks should not be used in ARS routing patterns.

Callback Queuing: Tie Trunk groups can be administered for Callback Queueing. If a user dials a Tie Trunk number and all trunks in the group are busy, the user must either wait for timeout or dial # to get queuing tone (if automatic queuing is administered) or reorder tone. Manual queuing can be activated after reorder tone is heard.

Conference: A tie trunk that is part of a conference counts as one of two allowable outside parties.

Direct Group Calling: Only automatic incoming tie trunks can be directed to a DGC group. However, dial-in tie trunks can access DGC groups.

Night Service: Dial-in tie trunks cannot serve as Night Service trunks.

Personal Lines: When a dial-in tie trunk is assigned as a Personal Line and the line is used for outgoing service at the same time that a call is coming in on the line, the terminal may be connected to the incoming call even though the call is intended for another terminal that shares the line. For this reason, it is recommended that tie trunks not be assigned as Personal Lines.

Administration Requirements

Trunk Port:

- Assign Trunk Type And Number.
- Assign Class of Service Code (0-15).
- Assign Facility Access Code; default = 102.
- Allow Dial Access (yes or no; default = yes).
- For Auto-in Type Only—Assign To DGC Group (Group Number 1-32, or 0 for none; default = 0).
- Make This a Directed Night Service Trunk (yes or no; default = yes).
- Assign Night Service Delay Announcement (1, 2, or 0 for none; default = 0).
- Dial-Inward Capability (Tone or Pulse; default = Pulse).

Port Options:

- Set E&M signaling type (Type I Compatible, Type V, Type I).

  System Administration must ensure that the port type (wink, delay or immediate service) and the signaling type (I Compatible, V, or I) is compatible with the distant PBX.

  — Port Type: Wink to Wink
  Delay to Delay
  Immediate to Immediate

  — Signaling Type: V to V
  I to I or I Compatible
  I Compatible to I or I Compatible

  When tie trunks are used with ARS (that is, the tie trunk is accessing the distant PBX WATS lines), wink or delay type trunks should be used. With immediate type trunks, the dialed digits may be spilled forward before the distant PBX is ready to receive them.

Hardware Requirements

Requires port interfaces on a TN760B Tie Trunk CP.

Touch-Tone and Dial Pulse Services

Description

All touch-tone single-line voice terminals and MET sets are equipped with dial pads that generate Dual Tone Multifrequency (DTMF) signals when a dial button is pressed. Model 500 Series single-line terminals have rotary dials that generate dial pulses corresponding to the numbers selected. The 7300H series (MERLIN) voice terminals are equipped with touch dial pads that generate digitally coded signals when a dial button is pressed.

Each trunk may be independently arranged for either touch-tone or dial pulse service.

Touch-Tone Dial Pads

On outgoing calls on trunks requiring touch-tone signals, cut-through-dialing is provided. Where the trunk requires dial pulse signals, conversion of the touch-tone signals to dial pulses is provided until an end of dialing signal is detected. Cut-through is then provided, and all subsequent digits are sent as touch-tone signals. See the “End-to-End Signaling” feature description for more information.

Rotary Dials

Voice terminals with rotary dials are required for Power Failure Transfer stations when the system connects to the CO by way of dial pulse trunks. They are not recommended for other applications.

Touch Dial Pads

For outgoing calls on trunks requiring touch-tone signals, all dialed digits are converted to touch-tone signals. If the trunk requires dial pulse signals, the dialed digits are converted to dial pulses until an end of dialing signal is detected. Cut-through is then provided, and all subsequent digits are converted to touch-tone signals. See the “End-to-End Signaling” feature description for more information.

Administration Requirements

Trunk Port:

- Assign trunk Class of Service (COS). COS includes touch-tone/dial pulse specification.

Transfer

Description

This feature allows a user to move any call from the user's terminal to another voice terminal, then disconnect from the call.

A user can transfer calls either with or without announcement. A multiline terminal user presses TRANSFER; the party is automatically placed on Special Hold (indicated by a broken-flutter on the status LED of the call appearance button). The system will automatically select an idle System Access button. The user may dial the desired number or select another facility button and dial the call. The user then can do one of two things: (1) hang up or (2) wait until the called party answers, announce the call, and then hang up. The held call receives Music-on-Hold, if provided and administered, until the transferring station hangs up, after which it receives ringback until the transferred-to station answers. Unanswered transfers will receive the coverage treatment of the transferred-to station.

A Personal Line transferred by a multiline voice terminal user will indicate the Special Hold status at the transferring voice terminal until answered, and may be reentered if the call is not answered. Reentering the call will automatically terminate the transfer attempt.

Single-line voice terminal users may transfer calls by flashing the switchhook, which puts the caller on hold, listening for Recall Dial tone, dialing the second party, and going on-hook either immediately or after announcing the call to the second party. A call may also be transferred by setting up a conference and then hanging up.

Considerations

Transfer provides a convenient way to redirect a call to another voice terminal. Attendant assistance is not required and the caller does not have to redial. While it is possible to transfer a call without announcement, it is recommended that call transfers be announced.

Interactions

The following features interact with Transfer.

Account Code Entry, Forced (FACE): Calls can be transferred in both directions between a FACE-restricted station and a non-FACE station.

Account *Code Entry, Optional:* A user may transfer a call to another user, then, instead of hanging up, enter an account code. Subsequent account code entries will be ignored.

Attendant Console: In most cases, the attendant should not use this feature to extend incoming calls, but should use the Start button or Selector Console instead. The exception to this rule occurs if a trunk-to-trunk transfer is desired (see below).

Bridging of System Access Buttons: A call can be transferred from a bridged call appearance using the usual transfer operations. When a call is held for transfer by pressing the TRANSFER button, an idle System Access button or an idle System Access-Originate Only button, if available, is automatically selected by the system for placing the new call. If neither of these appearance types is idle, the user can manually select a BA button or any other call button on which to place the new call.

The transfer operation and status indications of the principal's SA button (and its corresponding BA buttons) are similar to Personal Line operation with the following exceptions: Calls can be f rem/to an inside station or from/to an outside location, and after the transfer is completed (transferring station goes on-hook) the call will stay at the principal's SA button and its BA buttons only if one or more of these stations is bridged to the call. Otherwise, the call will be removed. An on-hold bridging station or principal is not considered to be bridged to the call.

Callback Queuing: Queued calls can be transferred. Single-line sets can transfer queued calls only before going on-hook. The transferring station must wait for the transferred-to facility to answer before completing the transfer; the transferred-to facility then receives queuing tone. Queued calls cannot be transferred to a tone (ringing, busy, etc.).

Calling Restrictions: A non-restricted user (typically the attendant) can transfer a CO trunk to an outward restricted or toll restricted station, giving the station outward service. The toll restriction class of the *transferring* station will apply for calls over a transferred trunk.

Coverage: Coverage treatment of the transferred-to station is provided to transferred calls. When a covering station transfers a covered call to another station, the call will no longer appear at the covering or the covered station. Note that if you attempt to transfer a call to a station that you provide coverage for, and that station does not answer, coverage might *not* be invoked. (This is one of the reasons why announced transfer is recommended.)

Display: At a station receiving a transferred call, the transfer descriptor "T" is displayed in position 1 before the transfer is completed. The transferring party's number and name are also shown.

Screen 1 | T785 Jones,B |

After the transfer is completed, "T" is removed and the display reverts to a standard incoming call format. Information about the transferred party is displayed.

Forwarding: Calls transferred by TRANSFER button operation to a forwarding station will be given normal Forwarding treatment.

Hold: An outside call placed on hold during call transfer receives music-on-hold, if available and administered. A user attempting to return to a held internal call that has been abandoned will hear nothing. A user attempting to return to a held CO trunk call that has been abandoned hears CO dial tone or receives CO intercept treatment until the CO disconnects.

Music-On-Hold: An administration item allows Music-On-Hold to be enabled or disabled for "Special Hold."

Park: Single-line voice terminals cannot transfer parked calls.

Pickup: A transferred call may be answered via the Pickup feature.

Remote Access: Remote Access callers cannot use the Transfer feature.

Trunk-To-Trunk Transfer: A trunk call may be transferred to another trunk. Refer to the "Trunk-To-Trunk Transfer" feature description for additional information.

Transfer To Data

Description

This feature allows multiline voice terminal users to originate or answer a call from their voice terminals and then establish a data connection by transferring the call to a data terminal.

Note, that single-line voice terminals can not be used to establish a data connection by transferring a call to a data endpoint.

(Refer also to the overview of the system's data features provided in the "Data Services Overview" description.)

Data terminal calls can be set up from a multiline voice terminal with a DATA button. The DATA button is associated by Data Dial Code (DDC) with a digital data endpoint. A separate DATA button must be provided for each data terminal that the voice terminal can transfer calls to.

The DATA button status LED provides status indications for the data endpoint:

- Dark—Data endpoint is idle
- Winking—Data endpoint is reserved
- Flashing —Data endpoint is being alerted to an incoming call
- Steadily Lighted— Data endpoint off-hook (busy).

The DATA button status LED will wink only when a voice terminal reserves a data endpoint by Data Call Preindication.

Data Call Origination Using Transfer to Data

A voice terminal user, after calling a DDC or a PDC (to reach an analog data endpoint) receives either answer tone or called party answer, respectively. The user then transfers the call to the associated data terminal by pressing DATA and hanging up. The called party may also use Transfer To Data to transfer the call to a data terminal, or may press the Data button on an associated modem.

An inside call cannot be transferred via Transfer To Data until the far end answers.

If a handshake failure occurs after Transfer To Data, the data call will be disconnected and the data terminal left in the idle (on-hook) state.

Note: Even if the associated data port is optioned for autobaud, the call will be set up at the highest common speed that the calling and called data terminals are administered for, independent of the current *data terminal* settings.

Data Call Preindication

A multiline voice terminal user, by going off-hook and pressing an idle DATA button, may indicate that a data call will be attempted. This reserves the associated data port and a modem pool conversion resource. This procedure is recommended when the data call is a trunk call. The data port reservation is acknowledged by a winking status LED at the DATA button. Subsequently, invoking Transfer To Data transfers the call to the associated data terminal.

Preindication is canceled:

- If the user goes on-hook before transferring the call to data
- If the user preindicates on a second DATA button
- If, after dialing is complete, a second DATA button is pressed. Preindication is canceled for the first data terminal and the data call is transferred to the second data terminal.

When Preindication is canceled, the associated pooled modem conversion resource reservation is canceled.

Interactions

The following feature interacts with Transfer To Data.

Modem Pooling: If a conversion resource is required on an external call, invoking Data Call Preindication will cause a pooled modem conversion resource to be reserved. If none is available (e.g., the system has no Pooled Modem CP), Reorder Tone is provided. (This will occur whether a conversion resource is actually required or not.)

Administration Requirements

Voice Terminal Port:

- Assign DATA button

Trunk Groups

Description

This feature allows each trunk in the system to be assigned to one of up to 16 trunk groups.

Trunks link two switching systems, such as System 25 and the local CO or System 25 and another PBX. Although not required, trunks can be grouped together in trunk groups (sometimes referred to as pooled facilities) when all the trunks in the group perform the same function. This grouping provides resource pooling that results in better service with a given number of trunks. It also simplifies administration and calling. Calls are routed to the appropriate trunk group; an idle trunk, if available, is selected from the group. Up to 16 trunk groups (pooled facilities) may be assigned in the system.

Several different kinds of trunk groups can be assigned in System 25:

- Central Office (CO)—Provides a link with the local CO for calls except Direct Inward Dial (DID) calls. Trunks classed as “CO” have a number of special characteristics.
- Foreign Exchange (FX)—CO trunks that connect to a CO other than the local CO.
- Wide Area Telecommunications Service (WATS)—CO trunks that connect to an Outward WATS office or a dial 800 (in-WATS) Service office.
- Direct Inward Dial (DID)—Provides incoming (only) service from the local CO. These calls go directly to voice terminals instead of through the attendant.
- Tie—Provides a link with another private switching system or network.

Trunk groups can be one-way (incoming) or two-way. Selection of the trunk group to be used for a given call is determined by the initial digits of a dialed number (or by the ARS feature). These digits are referred to as the facility access code. Each trunk group is assigned a unique code. Assuming an idle trunk in the selected group is found, a seizure signal (service request) is sent to the distant switch. If the distant switch requires dialed digits (as all but some tie trunks do), a signal (Dial Tone) is returned to System 25, indicating readiness to accept dialed digits.

Trunk type refers to the physical design of a trunk circuit. The trunk types supported and a brief description of each are given below. Refer to Section 3, “Functional Description” and Section 9, “Glossary” for additional information.

- Loop Start—A closure signal is sent through the loop formed by the trunk leads.
- Ground Start—Similar to loop start but enhanced with ground signals.
- Immediate Start—No start dial signals are used. On outgoing calls, the system waits at least 80 ms after sending the seizure signal before sending the digits required by the distant switch. This allows the distant switch enough time to attach a digit receiver to the trunk (Tie and DID trunks).

- Wink Start—A momentary signal (wink) is sent to the distant switch (Tie and DID trunks).
- Delay Dial—A steady signal is sent to the distant switch and is removed when ready to receive digits (Tie trunks only).
- Automatic—No start dial signals are used. The seizure signal sent or received is sufficient to route the call (Tie trunks only).

Trunk groups connecting with a local CO, WATS office, or FX office can be ground or loop start. DID trunk groups can be immediate or wink start. Tie trunks groups can be automatic, wink start, immediate start, or delay dial.

Dual Tone Multifrequency (DTMF) signaling (touch-tone) or dial pulse signaling can be used between the System 25 and the far end switch. System 25 can send or receive either type of signaling required by the distant switch (DID trunks can only receive dial pulse signals). The type to be used is specified when the associated trunk is administered.

An incoming call can be connected to another trunk, a voice terminal, a data endpoint, an attendant console, or an announcement. When the call is answered, an off-hook indication is sent to the serving office. This signal may be used to initiate the recording of call details normally used for billing.

Trunks in a two-way trunk group should be translated (at the SAT) in the same order that the serving office hunts when searching for an idle trunk. System 25 will then hunt in reverse order. This reduces the probability that both switches will attempt to seize the same trunk at the same time.

Considerations

Trunks of the same type and Class Of Service may be assigned a (Pooled) Facility Access Code. This provides users with dial or direct (button) access to the trunk pool. Trunks may be dial access restricted to reserve them for ARS and direct access only.

Refer to “Recommended Central Office Trunk Facilities” (Section 5) for an estimate of CO trunk requirements based on traffic considerations. See the “Pooled Facility-Direct Access” and “Pooled Facility-Dial Access” feature descriptions for additional information.

Trunks may be reserved for incoming calls (e.g., sales or service department calls) by specifying this in the (administered) Class of Service code.

Interactions

The following features interact with Trunk Groups.

Direct Group Calling: Most trunks may be administered so that incoming calls are directed to a specified DGC group. For tie trunks, only the automatic-in type may be so administered.

Facility Access Restriction (see "Calling Restrictions"): Stations may be restricted from dialing the CO trunk pool and/or all (fifteen) other trunk groups (as a whole). Stations so restricted may still dial out if they are transferred to a trunk by another station not so restricted.

Tie Trunks: Refer to the "Tie Trunks" feature description.

Toll Restriction (see "Calling Restrictions"): When toll restricted stations access FX, WATS, or Tie trunks, they *are not toll restricted* (i.e., toll restriction applies to CO trunks only).

Administration Requirements

Trunk Port:

- Assign Trunk Type And Number.
- Assign Class Of Service Code (DID: 1-4; Other: 0-15).
- Assign Facility Access Code; default codes are based on the CPS in a system. They are assigned as follows:

  Loop Start Trunks -100.

  Ground Start Trunks -101.

  Tie Trunks -102.
- Allow Dial Access (yes or no; default = yes).
- Assign To DGC Group (Group Number 1-32, or 0 for none; default = 0).
- Make This a Directed Night Service Trunk (yes or no; default = yes).
- Assign Night Service Delay Announcement (Announcement 1 or 2 or 0 for none; default = 0).
- Dial-Inward Capability (Tone or Pulse; default = Pulse [Tie trunks only]).

Hardware Requirements

Associated trunk port interfaces.

Trunk-To-Trunk Transfer

Description

This feature allows users to connect incoming trunk calls to other outside trunks and then hang up (under certain conditions).

Incoming trunk calls may retransferred to another trunk, or conferenced with another trunk. In all cases and at all times, either a System 25 station must remain in the conference or one of the calls must be an incoming call on a ground start, loop start (administered for trunk-to-trunk transfer), DID, or tie trunk. The other call may be on any type of trunk and may be incoming or outgoing.

Considerations

Trunk-to-trunk transfer is particularly useful when an outside caller requests a transfer to another outside number. For example, an employee can call in and have their call transferred elsewhere. Note that as long as an inside station stays on the call (even if a *multiline* station puts the call on hold and hangs up) *any* two trunks may be conference. If the station drops out of the call, the trunk conference will be torn down *unless* the above conditions are met.

If a System 25 station enters a trunk-to-trunk transfer call via a line appearance button for one of the conference trunks, the call will still be broken down when one of the outside parties hangs up.

A Direct Group Calling call that comes in on a ground start trunk and is answered at a single-line set is not eligible for trunk-to-trunk transfer.

Interactions

The following feature interacts with Trunk-To-Transfer Transfer.

Conference: Trunk-To-Trunk transfers may be set up using the Conference feature. The conference must include an incoming trunk call on either a ground start, DID, or tie trunk if it is to continue after all inside stations have dropped off.

Administration Requirements

System:

- Allow trunk-to-trunk transfer on loop start trunks? (yes or no; default = no).

  This capability should be assigned only where Central Offices give a reliable disconnect signal of at least 600 milliseconds at the end of the Loop Start call.

User Changeable Options

Description

User Changeable Options allows a data terminal user who is in the Command Mode to view and change the settings of certain data port options. This feature is available to users of Data Line CP ports; users of STARLAN Interface CP ports cannot change their port options. Table 2-L contains brief descriptions of the user changeable options.

Table 2-L. User Changeable Options

Option	Definition
Speed	low, 300, 1200,2400,4800,9600, 19200, autobaud
Parity	odd, even
Permit Mismatch	Allows two data endpoints to communicate at different rates.
Local Echo	Determines whether characters from the data equipment will be echoed by System 25 during Command Mode.
Answer Text	Enables call progress messages to be displayed at the called data endpoint.
Connection Indication	Determines whether users who have Command Mode enabled will receive the “CONNECTED” message when a connection has been made.
Recall Sequence (disconnect)	Two short breaks or one long break; the sequence used to disconnect a data call.

Note: The System Administrator may, under data port administration, deny permission for users of specific data ports to self-administer these options.

The user selects the Options menu from the Command Mode entry level menu. (Figure 2-55 illustrates all available Command Mode menus.) The user now has the choice of viewing options, changing options, or exiting the Options menu.

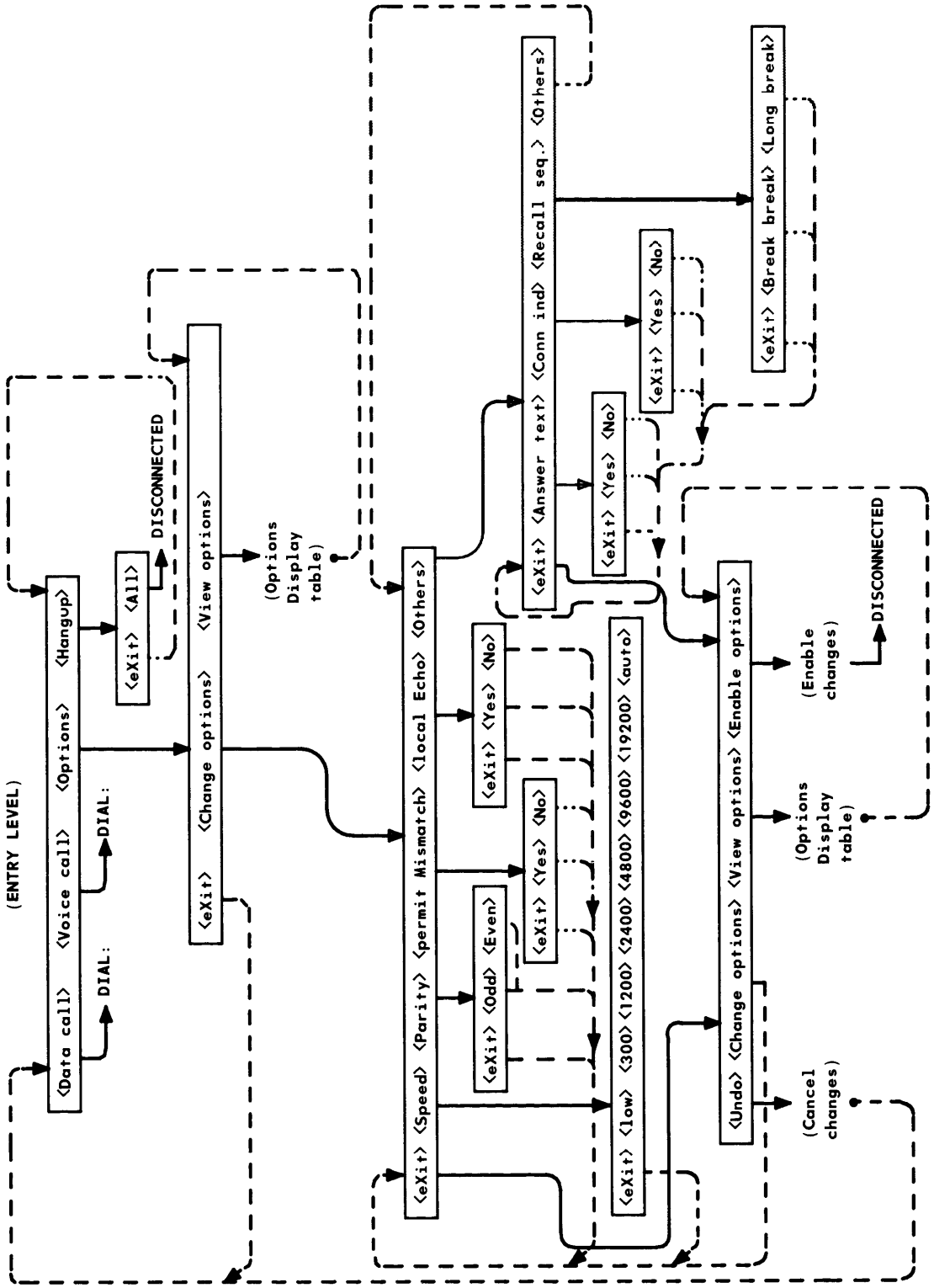


Figure 2-55. Command Mode Menu Tree

Viewing Options

When **<View options>** is selected, current and default values for the various data port options are displayed, as shown below:

OPTIONS	CURRENT	DEFAULT
Speed (highest)	9600	19200 (Auto)
Parity	Even	Even
Mismatch	Yes	No
Local Echo	Yes	Yes
Answer Text	Yes	Yes
Connect Indicat.	Yes	Yes
Recall Sequence	Br-Br	Br-Br

<eXit> <Change options> <View options>

At this point the user can exit from the View Options menu, Change options, or View options again (redisplays the Options table). If the user elects to exit, the terminal returns to the Command Mode entry level menu.

Note: Typing the capital letter found within a menu will *select* that item and move the user up or down the menu tree. For example, the user simply enters X or x (lower-case) to **<eXit>** the Options menu shown above and return to the entry level menu.

Changing Options—General

When on the Options Menu, the user selects **<Change options>** either by moving the cursor (with the space bar) beneath **<Change options>** and pressing RETURN, or by typing the single letter code (c) associated with that item.

If the user selects **<Change options>** from the Options menu, the first half of the Change Options menu is displayed, as shown below and on Figure 2-55.

```
<eXit>  <Speed>  <Parity>  <permit Mismatch>  <local Echo>  <Others>
```

If the user selects **<Others>,** the second half of the Change Options menu is displayed:

```
<eXit>  <Answer text>  <Conn ind>  <Recall seq. >  <Others>
```

If the user selects **<Others>** from the second half of the menu, the first half of the Change Options menu is redisplayed. In this way, users can “toggle” back and forth between the first and second halves of this menu.

Since these two lines are actually two halves of a single menu, users may select a particular menu item while active on either half of the menu. For example, users who are active on the first half of the menu may select **<Answer text>** by typing “a”.

Once a user has selected an option to be changed, a menu of valid settings for this option is displayed **(<Yes>, <No>,** etc.). An "X" is displayed beneath the current setting of the options, or beneath an option that may have been changed but not yet enabled. For all options except **<Speed>** (see below), settings may be selected either by moving the cursor (using the space bar) beneath the item desired and then pressing RETURN, or by typing the single-letter code associated with that setting. The user is then returned to the Change Options menu to make additional changes if required.

Changing Data Port Speed

The procedure for changing Speed settings is different from the procedure for changing the settings of other options. Within the Speed menu, the user may find that several values are marked with Xs. To change a speed, move the cursor beneath each value to be changed and type "+" to add the value or "-" to delete it. Once the new settings have been marked, press RETURN to translate the plus and minus signs to their proper "X" values and then type "x" to **<eXit>** from the Speed menu and return to the Change Options menu. For example:

The user enters the Speed menu and finds the following settings active:

```
<eXit>  <low>  <300>  <1200>  <2400>  <4800>  <9600>  <19200>  <auto>
                 X       X       X       X       X
```

To remove 9600 baud and activate autobaud, enter - under **<9600>** and + under **<auto>,** as shown below:

```
<eXit>  <low>  <300>  <1200>  <2400>  <4800>  <9600>  <19200>  <auto>
                 X       X       X       X       X-                +
```

With the cursor under any item except **<eXit>,** pressing RETURN provides the following:

```
<eXit>  <low>  <300>  <1200>  <2400>  <4800>  <9600>  <19200>  <auto>
                 X       X       X       X                         X
```

Entering "x" now allows the user to **<eXit>** the Speed menu and return to the Change Options menu. The user can make additional changes, as required.

When all of the changes have been made, the user should <eXit> the Change Options menu. The following menu is then displayed:

```
<Undo>  <Change options>  <View options>  <Enable options>
```

From the above menu:

- If the user selects **<Undo>,** the user is returned to the Command Mode entry level menu, deleting any option-change requests.
- If the user selects **<Change options>,** the Change Options menu is displayed and the user can make additional changes as required.

- If the user selects **<View options>,** the following menu is displayed:

OPTIONS	CURRENT	REQUESTED
Speed (highest)	9600	4800 (Auto)
Parity	Even	Even
Mismatch	Yes	Yes
Local Echo	Yes	Yes
Answer Text	Yes	Yes
Connect Indicat.	Yes	Yes
Recall Sequence	Br-Br	Br-Br

  The value in the CURRENT column indicates the current (active) status of the option. The value in the REQUESTED column indicates the most recently entered value (not yet enabled).

- If the user selects **<Enable options>,** the system incorporates the changes requested and displays the message **DISCONNECTED.** If Autobaud is off, the user must now press BREAK to return to Command Mode. If Autobaud is on, the user must press BREAK and RETURN to return to Command Mode.

Note: If a user attempts to enable options during a "SAVE" operation by the system administrator, the message "options changed FAILED" will be displayed. The user will be returned to the top level of the Command Mode menu, and all change requests discarded. After waiting a few minutes, the user may try again to change his/her data port parameters.

Considerations

For those cases where a data terminal user accesses various data endpoints, each requiring option changes, the User Changeable Options feature simplifies the process of administering the data port to allow data call setup. A system administrator is not required to enter each change.

Interactions

The following feature interacts with User Changeable Options.

Expert Mode: See the Expert Mode feature description for an abbreviated method of accessing Command Mode menus.

Administration Requirements

- Data Port:

  A data port must be administered to allow the terminal user to change options from the Command Mode menu. If permission is denied, the user may view the current option settings but not change them.

- AT&T STARLAN NETWORK Access:

  The User Changeable Options feature is not available for ports administered as STARLAN INTERFACE ports. Refer to the “STARLAN NETWORK Access” feature description for additional information.

Virtual Facilities

Description

A virtual facility (VF) is a call routing facility that is not defined by the physical facility (trunk) over which calls are routed, but is instead defined by a combination of access codes, authorization codes, and coded characters that allow special handling of the destination telephone numbers. VFs can be used to automatically route calls via other carrier networks, private networks, or tie trunks.

Up to ten virtual facility numbers (VFNs) may be administered. Each stored number may be up to 28 characters in length and is associated with a code in the range of #190 to #199. The first digits in a stored VFN must be the facility access code (FAC) for a physical trunk group over which the call is to be routed. A series of digits and special characters are stored following the FAC to define additional routes, Inter-Exchange Carrier (IXC) codes, identification codes, or instructions concerning special handling of the destination telephone number. When a VF has been defined using a particular trunk group, it has full access to all trunks in that group. It is considered "busy" only when the physical trunk group is busy. When a VF is dial accessed by a system user, calling restriction is based on the station's administered calling restrictions.

A system user may gain access to a VF by:

- Dialing the VF code (#190-#199).
- Storing the VF code as the first digits on a REP DIAL button. Other digits (for example, the destination telephone number) may be stored following the VF code.
- Dialing a System Speed Dialing code (#100-#189) that contains a VF as part of the stored number. The VF code must be the first digits stored in the Speed Dialing number.
- Dialing a Personal Speed Dialing code (#20-#39) that contains a VF as part of the stored number. The VF code must be the first digits stored in the Speed Dialing number.
- Using Automatic Route Selection (ARS) and having a VF as the route selected by ARS.

Note: The system can be administered to allow or restrict dial access for each VF code. If dial access is restricted (system default), a VF may be accessed only when used in an ARS routing pattern.

When virtual facilities are used in ARS patterns, they assume the same capabilities and restrictions as physical facilities. For example:

- Each VF may have a digit deletion and insertion scheme associated with it.
- Selective restriction of a VF may be accomplished by assignment of Facility Restriction Levels (FRLs).

If the VF is used in the first position of a routing pattern, calls may queue on it if all of the routes are busy.

Whenever a VF is used to complete a call (either by dial access or through ARS), call processing treats the number as a physical facility for Station Message Detail Recording (SMDR) purposes. Thus, if VF code #190 is used to complete a call, the SMDR call record will show "#190" as the facility used.

A VFN may contain up to 28 characters. The pound sign (#) is used as an escape character within the digit string and indicates that the character following the pound sign requires special interpretation. The following table defines the special characters that may be included in a VFN.

CHARACTER	FUNCTION
•	1.5 second pause
#*	Transmit *
# #	Transmit #
#5	Insert dialed digits (destination telephone number) here. The destination telephone number may be up to 16 digits in length (21 if ARS digit translations have occurred). If #5 *is* used, it must be placed within the last nine digits of the VFN. If #5 is *not* used within a VFN, the dialed digits are appended to the end of the VFN.
#8	Begin transmission of End-to-End Signaling (system begins transmitting touch-tone signals to the far end switch).

Examples of Virtual Facility Numbers

The use of VFs can be demonstrated with the following examples:

- **Example 1:** VF Code = #191, VFN = 10010288
  - — The first three digits (100) represent the FAC for a CO trunk group.
  - — 10288 represents an access code for a non-primary IXC.
  - — The destination telephone number (dialed by the user) will be transmitted after this IXC access code, since "#5" was not used within the VF number.
- **Example 2:** VF Code = #193, VFN = 2219*5554343*#5#*1 2345
  - — The first three digits (221) represent the FAC for a tie trunk to a remote PBX.

The “9” is used to access the remote PBX’s ARS.

— The “*” represents a pause of 1.5 seconds (allows time for dial tone to occur).

— The “555 4343” defines the local address of a private network and its internal routing table.

— The “*” represents a 1.5 second pause.

— The “#5” indicates that the destination number should be inserted here, rather than at the end of the VF translation.

— The “#*” indicates that the system should transmit a “*” symbol as the first character of an identification code.

— The “12345” represents the remaining characters in the identification code.

- **Example 3:** VF Code =#195, VFN = 1045554567’”1 234*9

This example demonstrates how a VF might be used in place of a tie trunk group connecting two local PBXs, when you have permission to access the other PBX’s facilities to complete calls.

— The first three digits (104) represent the FAC for a CO trunk group.

— The “5554567” represents the number for the other PBX.

— The “****” represents a 6 second pause (allows time for the other PBX to answer and return new dial tone).

— The “1234” represents a “barrier” (security) code required to access the other PBX’s facilities.

— The * represents a 1.5 second pause.

— The “9” represents an ARS access code for the other PBX.

— Since “#5” was not used within the VFN, the destination telephone number (dialed digits) will be transmitted after this ARS access code.

Accessing a Virtual Facility

Dial access is provided by dialing the VF code (#190-#1 99, including the "#"), followed by the destination telephone number.

- **Example:**

  — The user dials the following VF code and associated destination telephone number.

  #192 12125551643.

  — The stored VFN associated with VF code #192 is defined as 2222*333*444.

  — The first four digits (2222) represent the FAC for a tie trunk group to a remote PBX.

  — The * represents a 1.5 second pause.

  — The next three digits (333) represent the security code required by the remote PBX, indicating that you have permission to access their facilities.

  — The * represents a 1.5 second pause, as the remote PBX checks the validity of your security code.

  — The final three digits (444) represent the FAC required by the remote PBX to access their Band 5 WATS trunks.

  — Since "#5" was not embedded within this VFN, the destination telephone number (121 25551 643) will be transmitted after the WATS access code.

VF codes may be included in numbers stored in REP DIAL buttons, System Speed Dialing codes, and Personal Speed Dialing codes if the VF code is used at the beginning of these numbers. VF codes may not be assigned to FACILITY buttons and may not be embedded in other virtual facility numbers.

VFs may be used in ARS routing patterns just as if they were physical facilities.

Considerations

VFs enhance the Automatic Route Selection feature by increasing the number of facility types available for use in routing patterns. Using ARS ensures that the least-cost facility is used to complete each call. User intervention is minimized and associated user dialing errors are essentially eliminated.

In addition, in those systems where users are permitted dial access to VFs, user dialing of long digit strings is minimized, as are the associated dialing errors.

Interactions

When using a VF through dial access, calling restrictions will be based on the station's class of service. Dial access VF calls will be completed only if:

- The VF code is valid and not dial restricted.
- The station is not outward restricted.
- The station has dial access permission for the physical facility embedded within the VFN.
- The destination telephone number is valid and allowed for the station's toll restriction class.

VFs cannot be assigned to Facility buttons. Button access is provided by programming REP DIAL buttons only.

Administration Requirements

A VF must be programmed via System 25 administration. The following items are administrable:

- Specify a Virtual Facility code (#190-#199).
- Assign a Virtual Facility number to this code.
- Allow dial access to this Virtual Facility (yes or no).

Voice Message System

Description

The AT&T VOICE POWER Voice Message System (VMS) provides a group of communications services that expand System 25 operation in the area of collecting, processing, and delivering voice messages for inside users and outside callers. It functions somewhat like a sophisticated system-wide answering machine and/or an automated attendant. VMS is an adjunct to System 25, connected by way of special ports on analog tip/ring circuit packs. Administration procedures establish the proper System 25-to-VMS interface and set up the desired VMS capabilities.

The VMS hardware and software are part of System 25's Integrated Solution.

Five separate services are available with VMS. The first listed is user-oriented, that is, intended primarily for people inside the system. The other four are caller-oriented, that is, designed for managing incoming calls.

- Voice Mail Service—The primary service of VMS. It allows users to send voice messages to each other, retrieve their own messages (both inside and outside), record personal greetings to callers, and administer passwords.

- Automated Attendant Service—Answers incoming calls and gives the caller a choice of destinations (including attendant) or of recording a message for some inside station.

- Announcement Service—Provides simple announcements to callers, then disconnects. Useful for answering calls during nonbusiness hours and for providing information to employees.

- Call Coverage Service—Answers incoming calls and allows the caller to record a message or be transferred to an attendant.

- Message Drop Service—Allows certain users to record messages to solicit information from callers. After dialing an assigned number and hearing the message, callers can record their responses. Useful for product and marketing surveys.

For complete information on VMS, refer to its own set of documentation.

Interactions

The following features interact with VMS.

Leave Word Calling: VMS also supports the System 25 feature, Leave Word Calling (LWC), which is covered in a separate feature description in this section.

Night Service, Directed: VMS ports may be assigned Directed Night Service responsibilities.

Hardware Requirements

A Voice Message System must be connected to the System 25 by way of a port on a ZTN78 Tip Ring Line circuit pack. Use of a TN742 Analog Line circuit pack for VMS interface is not recommended.

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FUNCTIONAL DESCRIPTION

This section describes how the digital switch and the software of System 25 provide control and switching.

Digital Switch

Figure 3-1 shows a block diagram of the System 25 digital switch.

The basic switch hardware consists of the following:

- Common Control
- Switching Network
  - — Time Division Multiplex (TDM) Bus
  - — Port Circuits
  - — System Resource Circuits

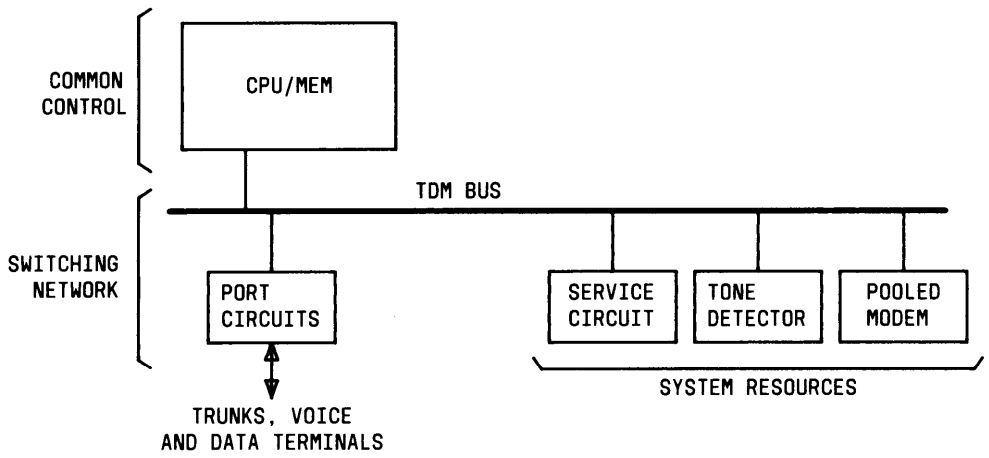


Figure 3-1. System 25 Digital Switch

Common Control

The Common Control circuitry consists of a single ZTN129 CPU/MEM (Call Processing Unit/Memory) circuit pack (CP).

CPU/MEM CP

The CPU/MEM runs the system feature code for the system and provides for the storage of software associated with system operation. This CP is powered from the backplane by +5 and -5 volts. It also draws -48 volts from the backplane to drive the Emergency Transfer Unit. The CP, shown in Figure 3-2, includes the following circuits:

- Microprocessor

  A 68010 16-bit microprocessor that executes call processing and data processing features. This includes all maintenance, administration, testing, and reporting software.

- Memory Management

  Memory management separates the on-board Random Access Memory (RAM) into 1024 memory pages of 256 bytes each. Each page is read and write protected and generates bus errors when violated.

- On-Board Memory

  On-board memory includes 1 M bytes of Read Only Memory (ROM) containing the power-up tests, the switch operating system, and the system operation software. In addition, there are 192k bytes of protected RAM containing writable data storage for call processing. The RAM is backed up by an on-board trickle-charge battery that maintains memory contents for up to two months. Of the 192k RAM, 32k is dedicated to translation data. The remainder is dedicated to call status data and the operating system message queues.

- EIA Channels

  Four asynchronous RS-232 EIA ports (1-4) are included to permit communication with an administration terminal, a Station Message Detail Recording (SMDR) device, and a digital tape unit. (The fourth port is reserved for future use.) Each port can support 300, 1200, 4800, or 9600 baud rates.

- Network Controller

  The network controller transmits control channel messages between the Call Processor and the port circuits over the TDM bus. The controller also monitors system clocks. The controller includes an 8-bit microprocessor that acts as a throttle passing messages between the Call Processor and the port board microprocessors.

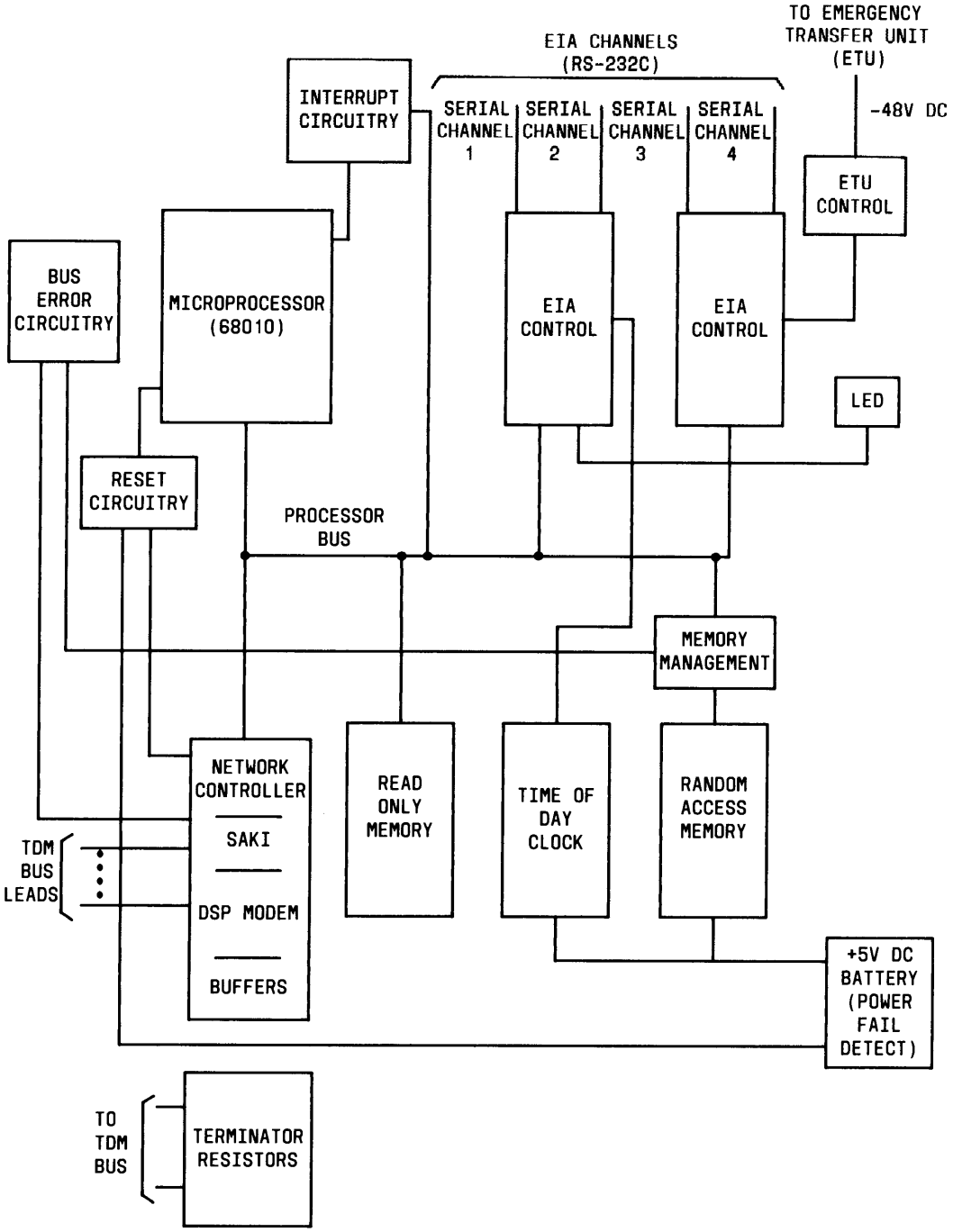


Figure 3-2. CPU/MEM (ZTN129) Circuitry

All uplink messages from the port circuits are checked for consistency and passed to the Common Control. The controller is the distribution control point for all downlink control messages. It continuously scans, over the TDM bus, the port circuit microprocessors for sanity and activity. External RAM associated with this microprocessor stores control channel information and port related information.

The controller consists of bus buffers and a Sanity and Control Interface (SAKI). It also contains a Digital Signal Processor (DSP) modem. The bus buffers provide the interface between the TDM bus and the on-board data buses to the SAKI and DSP modem. The SAKI receives and transmits control messages on the first five time slots on the TDM bus. The DSP modem is a 1200-baud, answer-only modem for Remote Initialization and Maintenance Service (RIMS) access. The microprocessor communicates with the SAKI, the DSP modem, and external RAM over the address and data bus.

- Clock

A clock provides both time-of-day information (in seconds, minutes, and hours), and the date to the 68010. The clock automatically adjusts for leap years. An on-board battery backs up the clock, so that accurate time is maintained even when the system power is off.

- Interrupt Circuitry

Interrupts are prioritized into seven levels, of which the highest (level 7) is nonmaskable. The interrupts are:

Interrupt	*Level*
AC Fail	7
Work cycle	6
Off board	5
EIA ports 3 and 4	4
EIA ports 1 and 2	3
Off board	2
Off board	1

- Reset Circuitry

The processor is automatically reset when power is turned on, when the +5 volt power supply drops below 4.5 volts (after it returns to +5 volts), or when the network controller determines that the processor is not functioning correctly. The processor can also reset the network controller when it determines that the network controller is not functioning correctly.

- Bus Error Circuitry

Bus errors suspend the processor from executing code. Bus errors are generated when memory management detects illegal reads or writes to RAM, when the processor attempts to access circuit packs or chips not physically present, or when the network controller determines that the processor is not functioning correctly.

- Emergency Transfer Unit (ETU) Control

  Removes -48 V dc power from the ETUs of the system when the system loses power or a major system malfunction occurs.

- Bus Terminators

  These resistors are required for proper operation of the TDM bus. The CPU/MEM CP provides the proper termination for one end of the bus, and a plug-in TDM bus termination circuit card (plugs into cabinet backplane) is used to terminate the other end. For this reason, the CPU/MEM CP must always be located in slot #1 of Cabinet 1.

Switching Network

System 25 uses distributed processing techniques to provide switched voice and data services. The switch operates at 64 Kbps. The switching network consists of the Time Division Multiplex (TDM) bus, the Port Circuits, and the System Resource Circuits.

The TDM bus connects the intelligent ports to the Common Control circuit pack and other ports through the network control circuit. The system resource circuits provide tone sources, receivers, detectors, and pooled modems. The intelligent ports connect external communications facilities to the TDM bus.

TDM Bus

The TDM bus consists of two groups of eight signal leads and five control leads, each with matching grounds. The port circuit packs place digitized voice [pulse code modulated (PCM)] signals on the bus.

The bus operates at 2.048 MHz. The framing pulse rate is 8 kHz. This provides 256 time slots (0-255) on the bus. The time slots are 488 ns wide. Time slots are generated as shown in Figure 3-3. The first five time slots are used for communications between the Common Control, the intelligent port, and resource circuit packs.

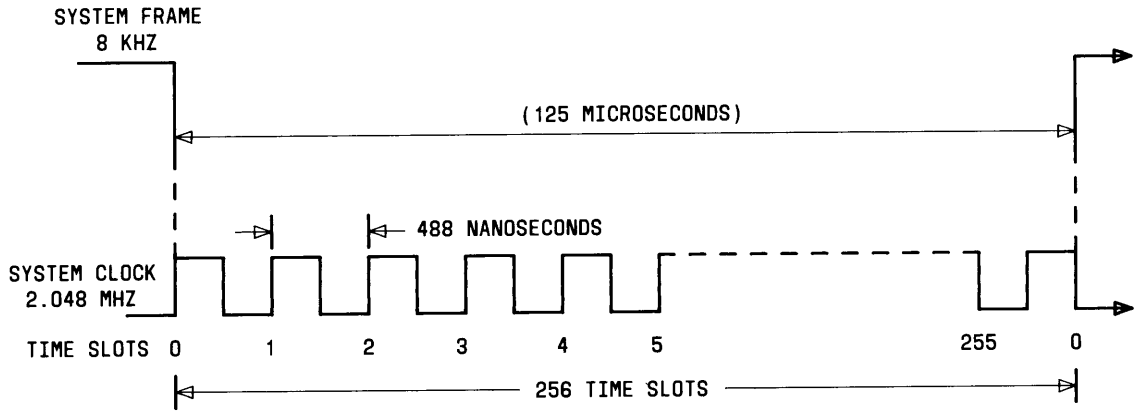


Figure 3-3. TDM Bus Time Slot Generation (Not A Timing Diagram)

Two time slots are required for each 2-party conversation. Each party transmits (talks) on one time slot and receives (listens) on another. Only five parties are allowed in a conference. During a conference connection, each member of the conference transmits on an individual time slot while receiving on as many as four other time slots. The actual switch capacity is 115 simultaneous 2-party conversations.

Table 3-A shows the allocation of the 256 time slots. Five are used for system control, 17 for tones, 232 for call processing, and two are not used.

- Physical Characteristics

  The TDM bus is an 8-bit bus that snakes continuously between cabinets in a multicabinet system as shown in Figure 3-4. The total length is about 9 feet for a three cabinet system. The bus is driven from any of the circuit packs in the cabinets. Similarly, a signal on the bus can be received by any circuit pack.

  Within a cabinet, the bus is printed on one side of the circuit pack carrier backplane while the other side is solid ground. Ribbon cables are used to connect the TDM bus between cabinets in a multi-cabinet system.

- Electrical Characteristics

  The TDM bus is an unbalanced, low characteristic impedance transmission line. Paths printed over a ground plane on the carriers and the flat ribbon cables between carriers maintain this impedance level over the full length of the bus.

  One end of the bus is terminated to ground with a bus termination circuit card and the other end is terminated by a network on the ZTN129 CPU/MEM CP. Each circuit pack connects to the bus through a custom bus driver device. The bus driver is a switchable constant current source so that even in the “high” output state there is no bus loading to cause reflections. The current output of the drivers is adjusted so that logic “high” is 1.5 volts compared to a “low” of 0 volts.

Table 3-A. TDM Bus Time Slots

Time Slot No.	Function	Time Slot No.	Function
00-04	Control (5)	15	941 Hz*
	Tones (17)	16	1209 Hz*
05	RIMS Listen	17	1336 Hz*
06	RIMS Talk	18	1447 Hz*
07	Dial Tone	19	1637 Hz*
08	Busy Tone	20	Data Null
09	Ringback Tone	21	Reorder Tone
10	Voice-Null	22-253	Flexible (232)
11	Music on Hold		
12	697 Hz*		
13	770 Hz*	254,255	Not used (2)
14	852 Hz*		

*These tones are used to generate touch-tone signals.

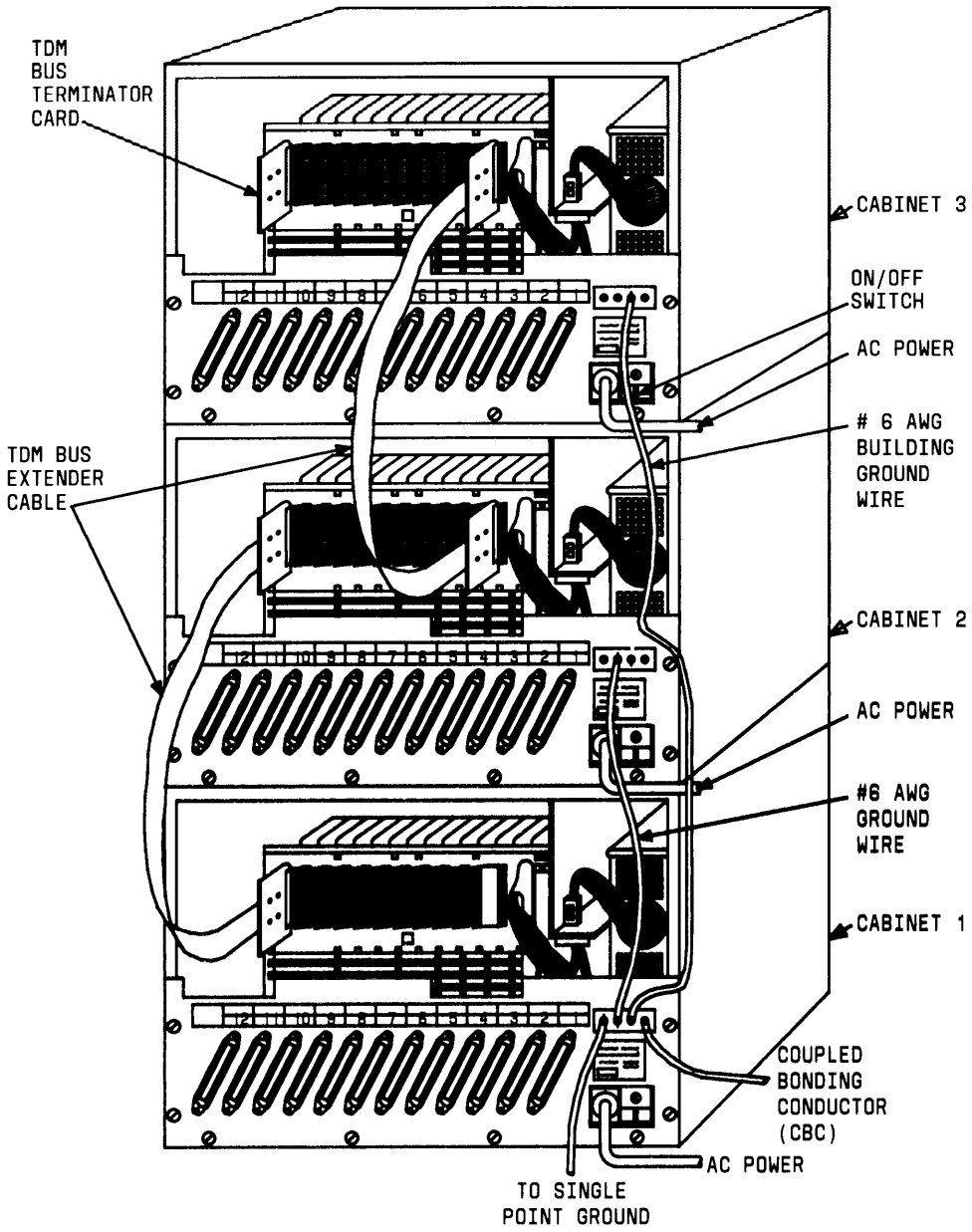


Figure 3-4. TDM Bus-Three Cabinet System

Port Circuits

The port circuit packs listed below provide the link between trunks and external equipment and the TDM bus. Figure 3-5 shows the equipment types that can be connected to the digital switch by the call processing and port circuit packs.

Analog Line (TN742)	Loop Start Trunk (ZTN77)
ATL Line (ZTN79)	MET Line (TN735)
Auxiliary Trunk (TN763)	STARLAN Interface (ZTN84)
Data Line (TN726)	Tie Trunk (TN760B)
DID Trunk (TN753)	Tip Ring Line (ZTN78)
Ground Start Trunk (ZTN76)	

Circuitry Common to All Port CPs

Eight port circuits are provided on most port circuit packs. The Multibutton Electronic Telephone (MET) Line, Tie Trunk, and Auxiliary Trunk Circuit Packs each contain four port circuits. The port circuits provide an interface between terminals/trunks and the TDM bus. The number of port circuit packs required varies according to customer requirements and equipment configuration. Each of the System 25 port circuit packs contain a number of common elements (see Figure 3-6) as well as the unique port circuits. The common elements are as follows:

- Bus Buffers

  The bus buffers are the digital interface between the backplane TDM bus wires (system bus) and the on-board circuitry (data bus). They also receive and distribute clock and frame signals.

- On-Board Microprocessor With External RAM

  The on-board processor does all low level functions such as scanning for changes and relay operations. In general, it carries out commands received from the Common Control and reports status changes to it. The external RAM stores control channel information and port-related information.

- SAKI (Sanity and Control Interface)

  The SAKI is the control interface between the Common Control that sends information via the network control circuit down the TDM buses and the on-board circuitry controlled by the on-board microprocessor. The SAKI receives control information (down-link messages) on the first five time slots and, as requested by the on-board microprocessor, transmits control information (up-link messages) on these same time slots. *(Continued on page 3-13.*)

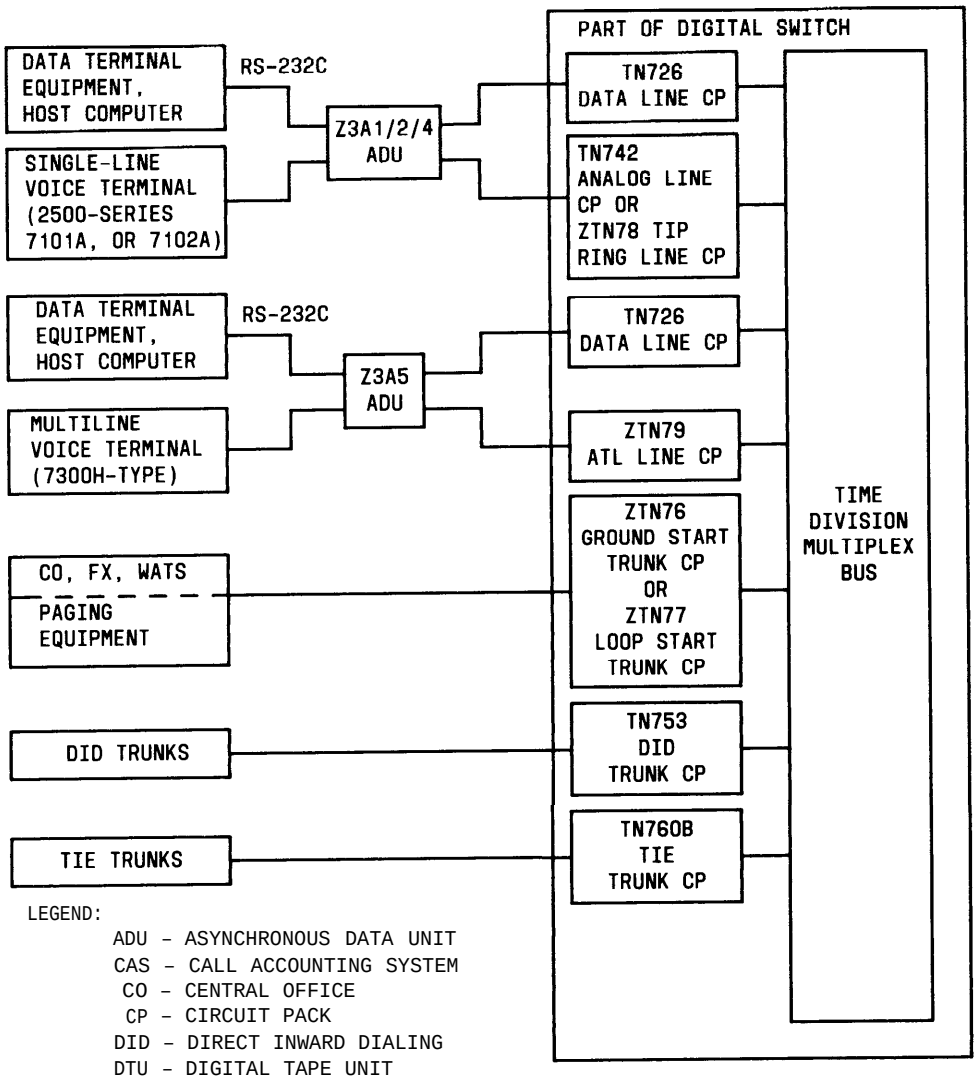


Figure 3-5. Equipment Connections Via Circuit Pack Ports (Sheet 1 of 3)

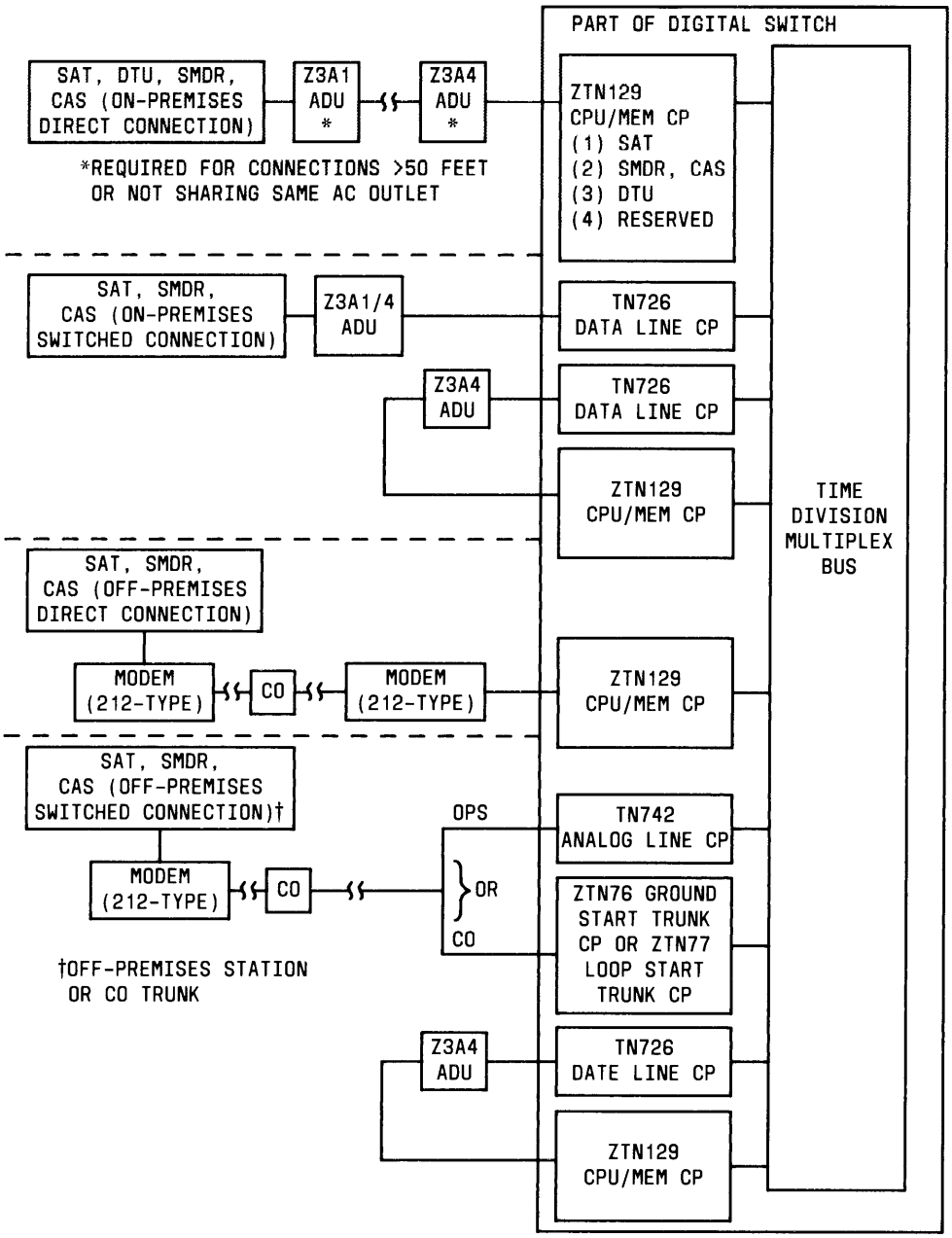


Figure 3-5. Equipment Connections Via Circuit Pack Ports (Sheet 2 of 3)

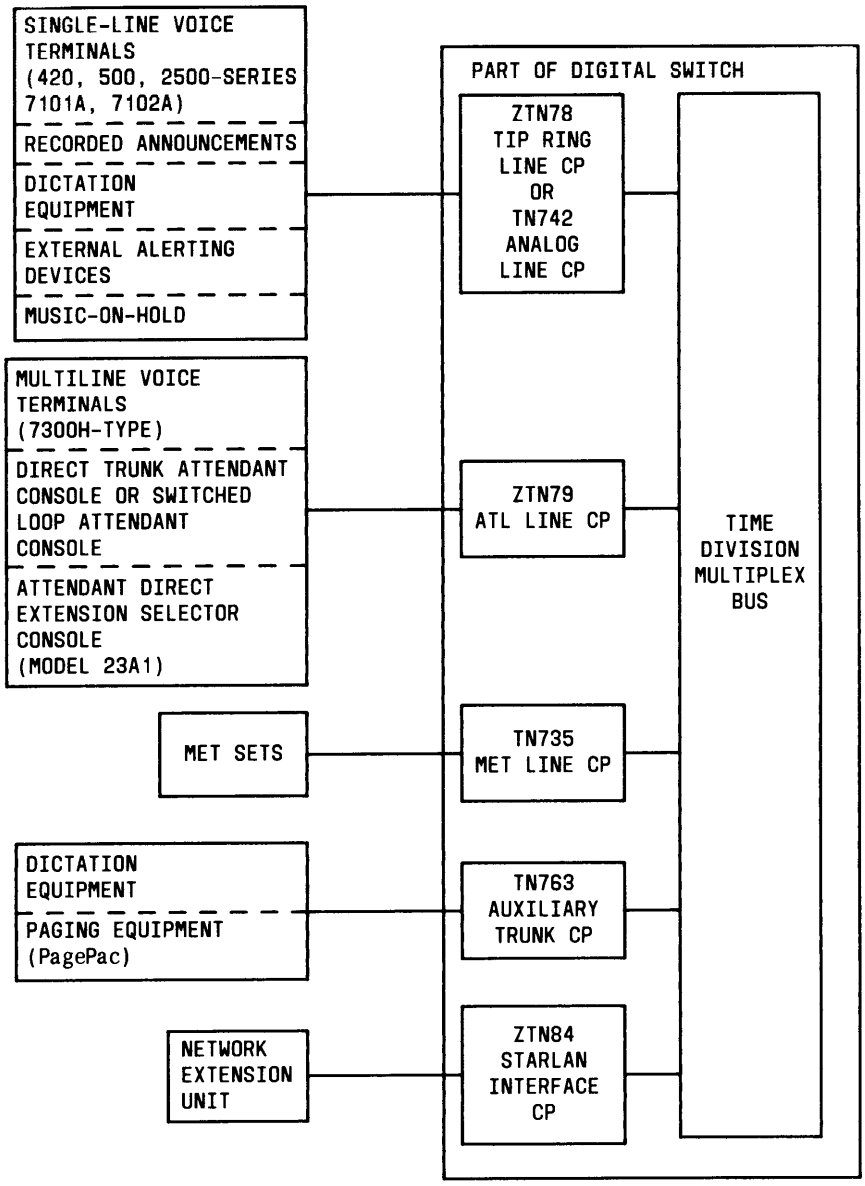


Figure 3-5. Equipment Connections Via Circuit Pack Ports (Sheet 3 of 3)

The SAKI also does the following functions:

- Identifies the circuit pack to the Common Control (location and vintage)
- Controls status indicator Light-Emitting Diodes (LEDs) — red (failure), green (translated), and yellow (circuit busy)
- Initiates power-on startup procedures
- Checks the on-board microprocessor for sanity and causes reinitialization in case of problems
- Takes NPEs out of service under control of the on-board microprocessor
- Resets the protocol handler on the ATL Line Circuit Pack
- Takes the whole circuit pack out of service on command from the Common Control or when it determines that on-board interference is present in the control time slots.

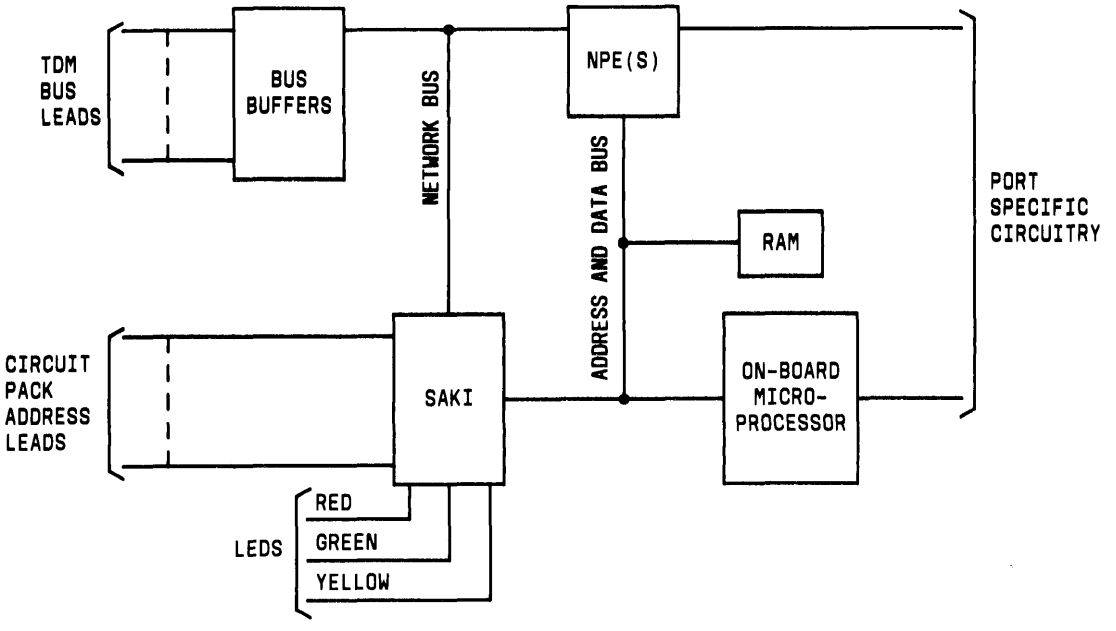


Figure 3-6. Port Circuit Pack Common Circuitry

- NPE (Network Processing Element)

  Each port circuit pack contains one or two NPEs. The Analog Line, ATL Line, Tip Ring, Data Line, Ground Start, Loop Start, and DID Trunk Circuit Packs contain two NPEs. The MET Line, Auxiliary Trunk, and Tie Trunk Circuit Packs contain one NPE.

  The NPEs do switching network functions for the port circuits. Under control of the on-board microprocessor, an NPE can connect a port circuit to any one of the TDM bus time slots. More specifically, it allows a port circuit to talk on one time slot and listen to the same time slot (NPE sidetone) and on up to four other time slots at the same time. In 2-wire circuits that provide their own sidetone, the NPE sidetone is not used.

- Circuit Pack Address Leads

  Seven leads (BA0-BA6) are tied to corresponding logic levels to uniquely identify each CP slot in the system, including multiple cabinet systems. The logic values on leads BA4 and BA5 are used to identify the cabinet (Cabinet 1, 2, or 3) and are tied via the cabinet address plugs to either +5 V dc or ground, as appropriate. Lead BA6 is tied to ground.

Analog Line (TN742)

The Analog Line CP (Figure 3-7) interfaces eight analog voice terminal lines and the TDM bus. The TN742 can be used instead of the ZTN78 Tip Ring CP. The TN742 supports up to five bridged single-line voice terminals, but only two can be off hook at one time. The ZTN78 CP does not support bridged terminals. In addition, the TN742 supports out-of-building, extended, and off-premises stations, the ZTN78 does not. The Analog Line has the following unique circuitry:

- Ringing Application Circuit

  This circuit receives ringing voltage from the power supply. It monitors ringing voltage and current, generates signals to the on-board microprocessor indicating zero ringing voltage and current, and detects a terminal user lifting the receiver during ringing. This prevents the application of ringing to the port circuit when a terminal user lifts the receiver during the ringing phase. Maintenance circuitry is also included. The maintenance circuitry detects when a terminal is connected to the port circuitry and checks for faults in the ringing application circuitry.

- Port I/O Circuit

  This circuit consists of bus expanders connecting the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

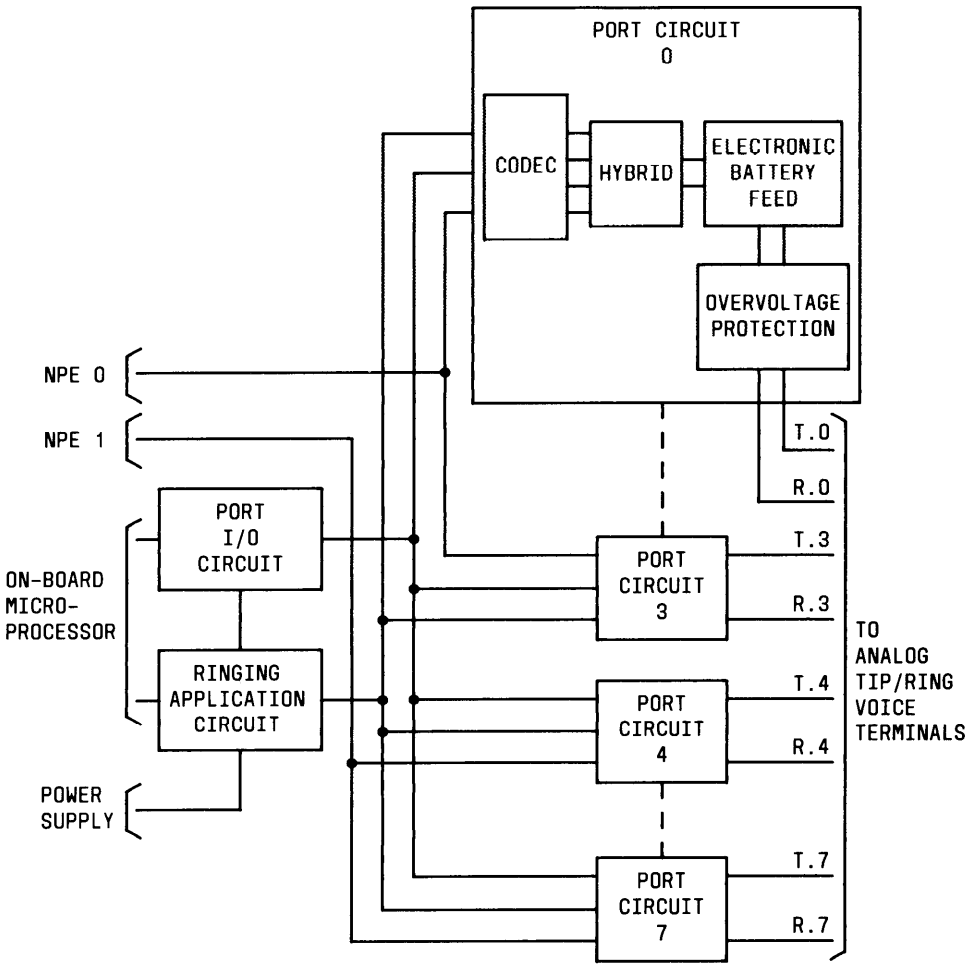


Figure 3-7. Analog Line (TN742) Unique Circuitry

- Port Circuits

  The eight port circuits are identical. Each port circuit consists of a coder/decoder (codec), hybrid circuit, electronic battery feed circuit, ring relay, and overvoltage surge protection circuit.

  The codec is a 4-wire circuit that converts the analog signal from a voice terminal to a PCM data signal. It converts an incoming PCM data signal from the NPEs to an analog signal. The hybrid circuit converts the 4-wire analog signal from the codec to a 2-wire analog signal that is connected to the analog line. Filtered power is provided for the codec and hybrid circuits.

The electronic battery feed circuit provides talking battery to the voice terminal. It also produces a controlled dc battery feed for short and long loops, detects when a receiver is lifted, and provides the message waiting signal by periodically turning off the feed voltage.

The ring relay provides the interface between the ringing application circuit and the port circuit. It causes ringing turn on and turn off.

The overvoltage surge protection circuit provides lightning surge and power line cross protection for the circuit pack.

ATL Line (ZTN79)

The ATL Line CP (Figure 3-8) interfaces eight hybrid voice terminal (7300H series) lines and the TDM bus. It terminates three pairs of wires from each terminal: analog voice pair, digital control pair, and power pair. The ATL Line has the following unique circuitry:

- Protocol Handler

  The 8-bit on-board microprocessor translates the control information in Control Channel Message Set (CCMS) message format to the control information message format used by the 7300H series voice terminals. The protocol handler sends the messages to the terminals via transceivers located in the port circuits.

- Port Circuits

  Each port circuit is identical. A port circuit consists of an analog port, one-half of a transceiver, and an electronic power feed device.

  The analog port circuit consists of a codec, a hybrid circuit, an isolation transformer, and associated power filtering circuitry. The codec and hybrid circuit perform the same function as the codec and hybrid circuit in the Analog Line Circuit Pack (TN742). The output of the hybrid circuit is connected to the primary of the isolation transformer. The secondary of the transformer is connected to the analog voice pair.

  The transceiver interfaces the voice terminal pair to the protocol handler. The electronic power feed device provides -24 volts dc on the power pair to the voice terminal. The device is polled by the on-board microprocessor, periodically and on demand, to test for an overcurrent or no-current condition.

  Each Electronic Power Feed (EPF) circuit supports two ports. If one of the associated lines becomes overloaded, the associated pair of lines will also be out of service. One EPF supports Ports 0 and 1, one Ports 2 and 3, one Ports 4 and 5, and one Ports 6 and 7. The on/off state of the device is controlled by the on-board microprocessor.

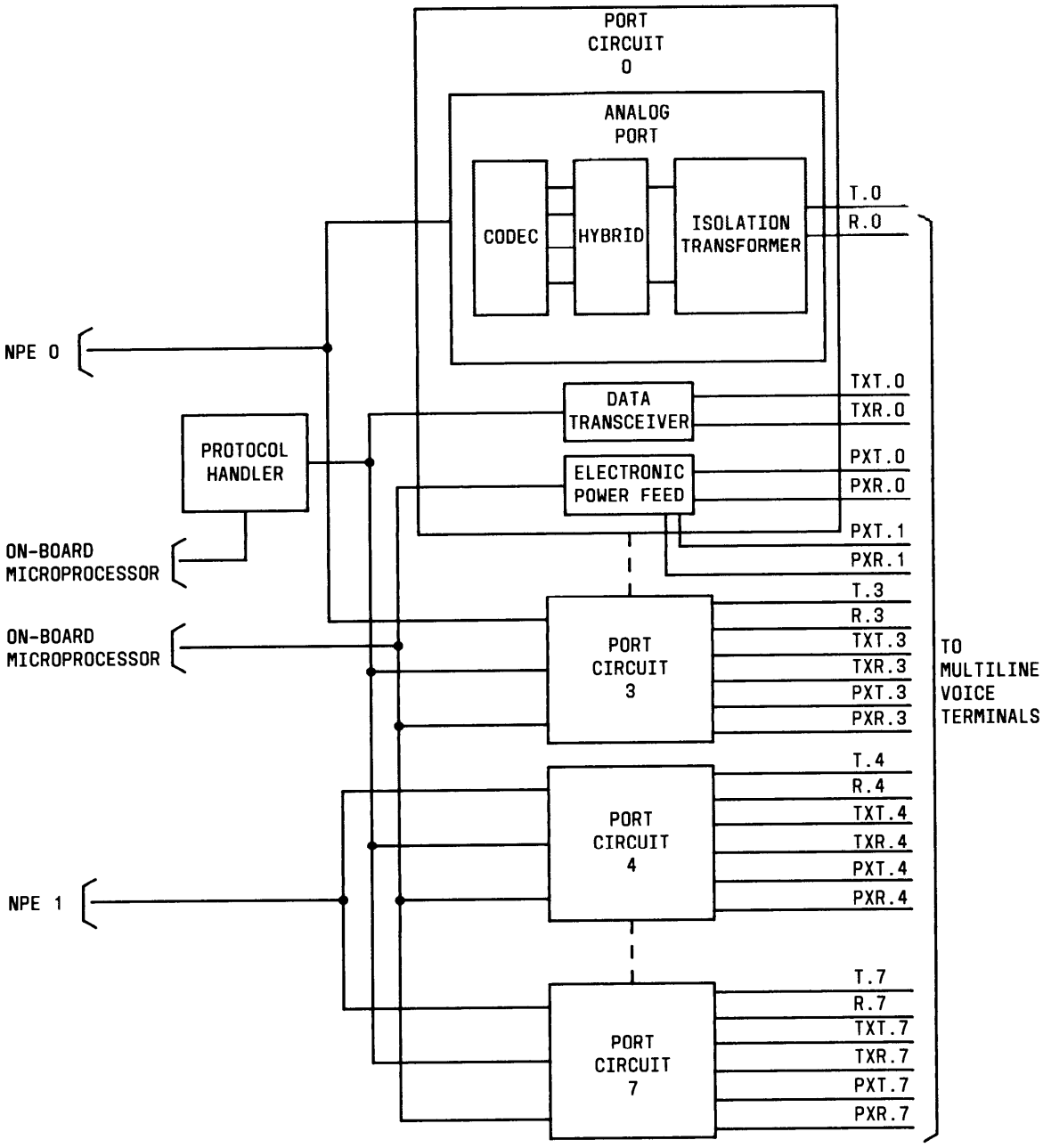


Figure 3-8. ATL Line (ZTN79) Unique Circuitry

Auxiliary Trunk (TN763)

The Auxiliary Trunk Circuit Pack (Figure 3-9) interfaces four ports provided for client-provided equipment (CPE) and the TDM bus. It is connected to the CPE by up to three pairs of wires. The transmission pair (T and R) carry voice signals and touch-tone control signals. The T and R also provide a loop start seizure indication to the CPE. The seizure pair (SZ and SZ1) provide seizure indication to the CPE. The signal pair (S and S1) provide answer supervision and/or make-busy information from the CPE. Depending on the application, either the transmission pair only or all three pairs are connected to the CPE. The Auxiliary Trunk has the following unique circuitry:

- Ground Detector Circuit

  This circuit determines if an answer-supervision or make-busy signal from the CPE is present. The inputs of the ground detector come from the port circuits as an analog current to the -48 volt dc supply. Its output is a port control point to the port I/O circuit.

- Port I/O Circuit

  This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

- Port Circuits

  The four port circuits are identical. Each port circuit consists of a codec, hybrid circuit, line transformer, relay driver, battery polarity sensor, and surge protection circuit.

  The codec is a 4-wire circuit that converts the analog signal from the CPE to a PCM data signal. It converts an incoming PCM data signal from the NPE to an analog signal. The hybrid circuit converts the 4-wire analog signal from the codec to a 2-wire analog signal that is connected to the CPE by a line transformer.

  The relay driver buffers and inverts the relay drive signals from the port 1/0 circuit so that a logic high input operates the appropriate relay. The relays control circuitry that provide the proper interfaces for CPE.

  The surge protection circuit provides lightning surge protection for the circuit pack.

  The circuit pack supports both touch-tone and dial pulse signaling. Longitudinal surges are isolated from the hybrid and codec by the line transformer.

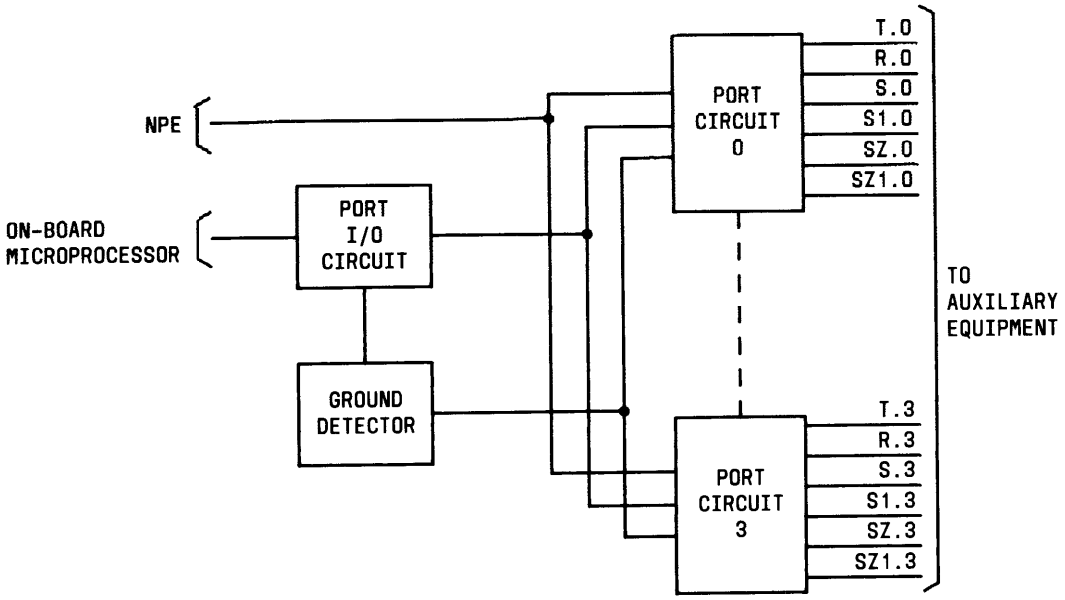


Figure 3-9. Auxiliary Trunk (TN763) Unique Circuitry

Data Line (TN726)

The Data Line Circuit Pack (Figure 3-10) interfaces eight Asynchronous Data Units (ADUs) data devices and the TDM bus. The ADUs are typically, in turn, connected to RS-232 type devices. The Data Line has the following unique circuitry:

- Bit Clock

  The bit clock circuitry is used to provide the Octal Asynchronous Terminal Mode Two EIA Asynchronous LSIs (OATMEALs) with a clock frequency that is a multiple of each baud rate. In addition, the clock rate is divided down to 160 kHz. The 160 kHz is then compared to the 160 kHz system data clock and is phase-locked to it. The phase-locked circuit is required for low speed operation.

- Bus Isolation

  This portion of the circuit pack is used to isolate the microprocessor bus. Isolation is required because the realized bus load exceeds the maximum limit specified for this device, due to the large number of devices controlled by the NPE. The OATMEALs are isolated from the common bus structure.

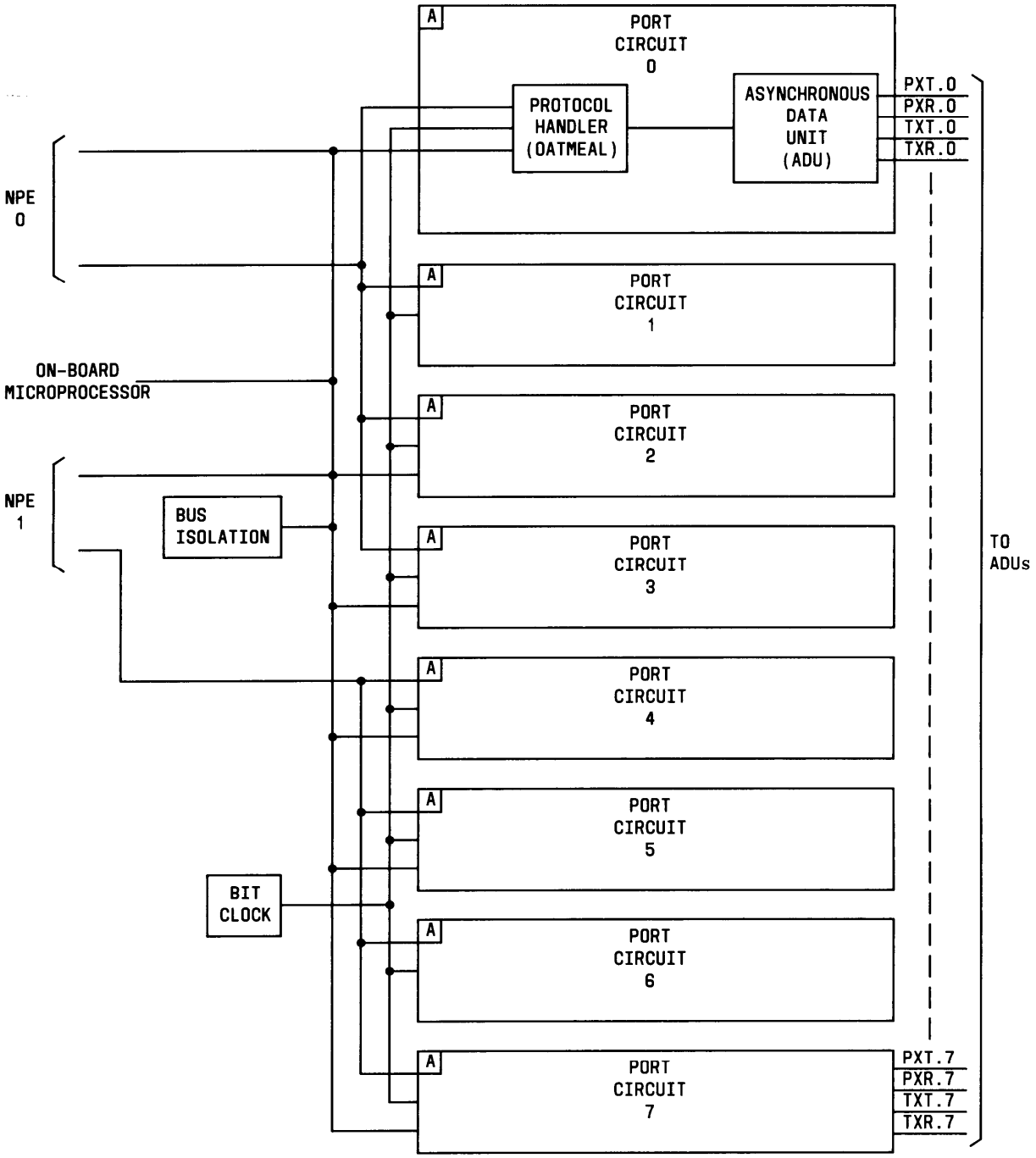


Figure 3-10. Data Line (TN726) Unique Circuitry

- Port Circuits

  Each of the eight identical port circuits allows the connection of interface equipment having an RS-232 compatible serial interface to the switch. The circuit provides asynchronous full duplex data transport at standard speeds from 300 to 19,200 bps and a low data rate (<300 bps). Each port includes an Asynchronous Data Unit (ADU) to extend the serial communications link length and provide safe isolation. The ADU terminates to another ADU at the Customer Provided Equipment (CPE). The distance between the digital switch and CPE is inversely proportional to the speed at which the link is run.

  Throughout the circuit, various gates are used to provide a means of isolating devices for automated circuit pack testing. Typically, these devices are crystal oscillators or memory components attached to the microprocessor bus.

DID Trunk (TN753)

The DID Trunk CP (Figure 3-11) interfaces eight central office trunks arranged for Direct Inward Dialing (DID) and the TDM bus. The DID Trunk has the following unique circuitry:

- Port I/O Circuit

  This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

- Port Circuits

  The eight port circuits are identical. Each port circuit consists of a codec, balance network, trunk interface unit, and loop termination circuit.

  The codec is a 4-wire circuit that converts the NPEs output to an analog signal. Likewise, it converts the analog signal from the Central Office (CO) to a PCM signal to the NPE.

  The trunk interface unit contains a hybrid, a 2-wire interface circuit, and control circuitry. The hybrid circuit converts the 4-wire analog signal from the codec to a 2-wire analog signal that is connected to the analog line by the 2-wire interface circuit. The control circuitry controls loop current, internal signal gain, terminating resistance, battery feed shutdown, and battery reversal. The circuit pack accepts dial pulse signaling.

  The loop termination circuit provides a fixed impedance to the DID trunk.

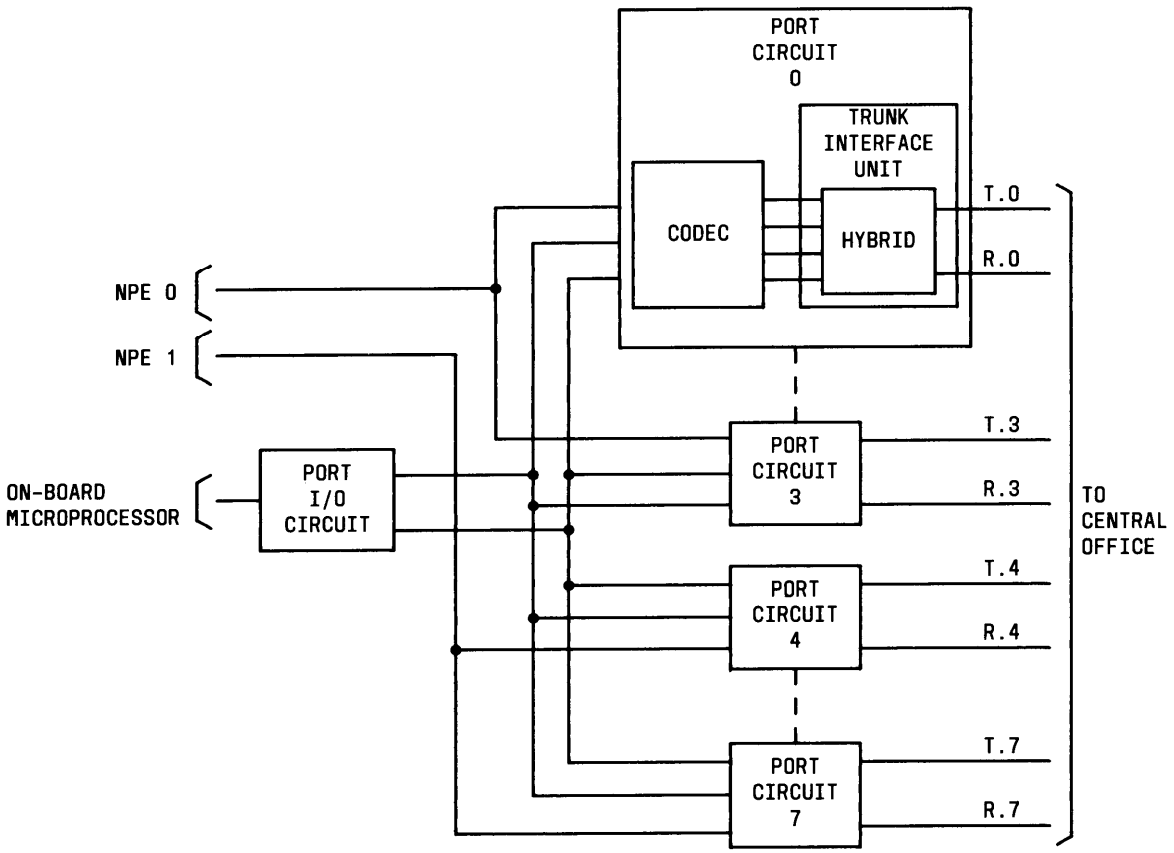


Figure 3-11. DID Trunk (TN753) Unique Circuitry

Ground Start Trunk (ZTN76)

The Ground Start Trunk CP (Figure 3-12) interfaces eight central office trunks and the TDM bus. The Ground Start Trunk has the following unique circuitry:

- Ground Detector Circuit

  The ground detector circuit determines if ground has been applied to the tip lead for incoming seizure. It also senses tip ground on outgoing seizure indicating dial tone is present. One ground sensor is used for each port circuit. Input for the ground sensor comes from the port circuit as an analog current to the -48 volt dc supply. The output of the ground sensor is a port control point to the port I/O circuit.

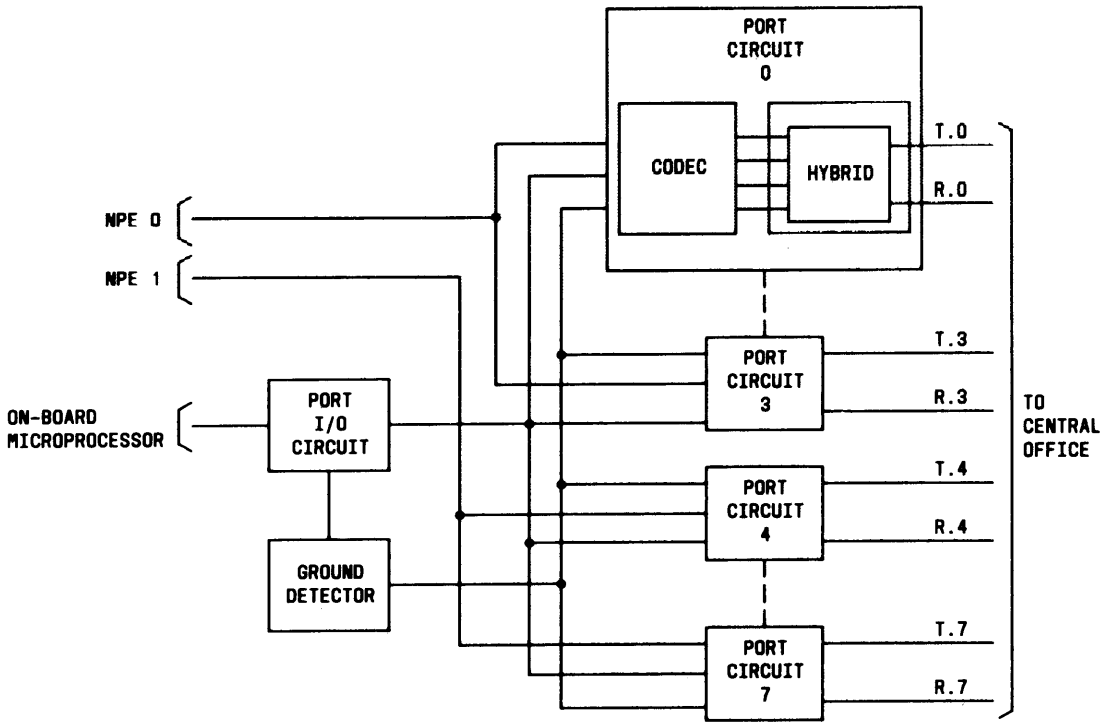


Figure 3-12. Ground Start Trunk (ZTN76) Unique Circuitry

- Port I/O Circuit

  This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

- Port Circuits

  The eight port circuits are identical. Each port circuit consists of a coder/decoder (codec), hybrid circuit, line transformer, relay driver, and surge protection circuit.

  The codec is a 4-wire circuit that converts the NPEs digital output to an analog signal. Likewise, it converts the analog signal from a central office trunk to a Pulse Code Modulated (PCM) data signal to the NPE. The hybrid circuit converts the codec 4-wire analog signal to a 2-wire analog signal that is connected to the central office trunk by the line transformer.

  The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input operates the appropriate relay. The relays control circuitry provides the proper signaling for ground start trunks. The trunks support touch-tone dialing. The surge protection circuit provides overvoltage lightning surge protection.

Loop Start Trunk (ZTN77)

The Loop Start Trunk Circuit Pack interfaces eight central office loop start trunks and the TDM bus. Figure 3-13 shows the Loop Start Trunk unique circuitry.

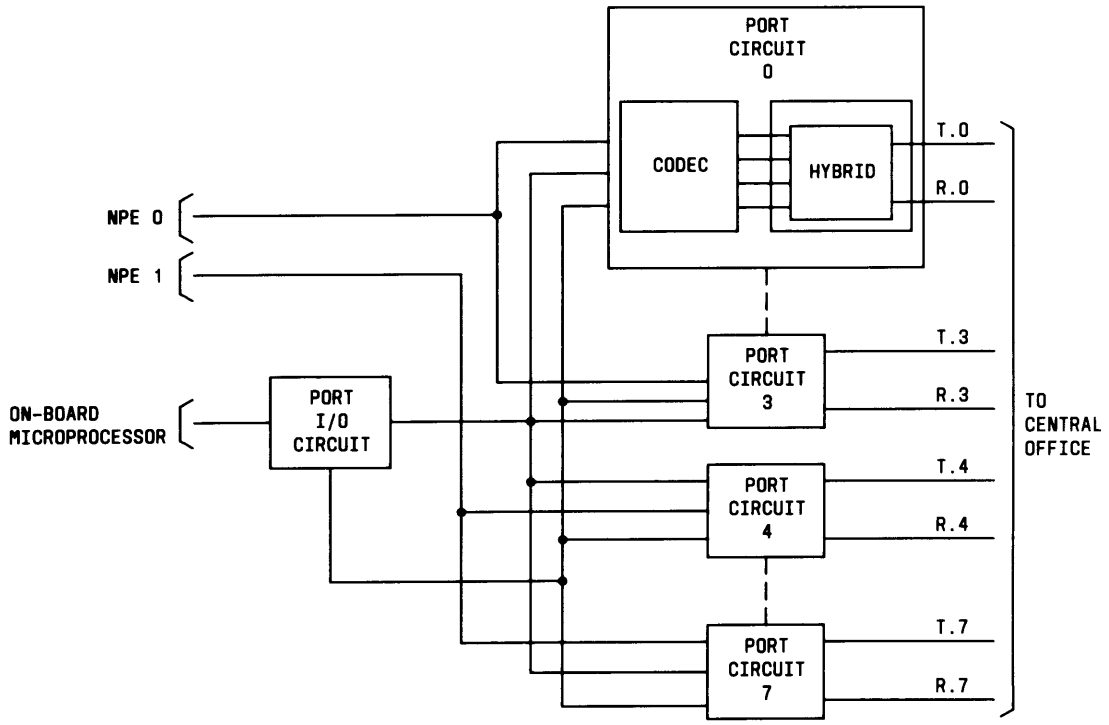


Figure 3-13. Loop Start Trunk (ZTN77) Unique Circuitry

- Port I/O Circuit

  This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

- Port Circuits

  The eight port circuits are identical. Each port circuit consists of a codec, hybrid circuit, line transformer, relay driver, and surge protection circuit. The codec is a 4-wire circuit that converts the NPEs output to an analog signal. Likewise, it converts the analog signal from a central office trunk to a PCM data signal to the NPE. The hybrid circuit converts the codec 4-wire analog signal to a 2-wire analog signal that is connected to the central office trunk by the line transformer. The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input

operates the appropriate relay. The relays control circuitry provides the proper signaling for loop start trunks. The trunks support touch-tone dialing and dial pulse signaling. The surge protection circuit provides overvoltage lightning surge protection for the circuit pack.

MET Line (TN735)

The MET Line Circuit Pack interfaces four Multibutton Electronic Telephone (MET) lines and the TDM bus. The MET Line unique circuitry consists of four port circuits as shown in Figure 3-14.

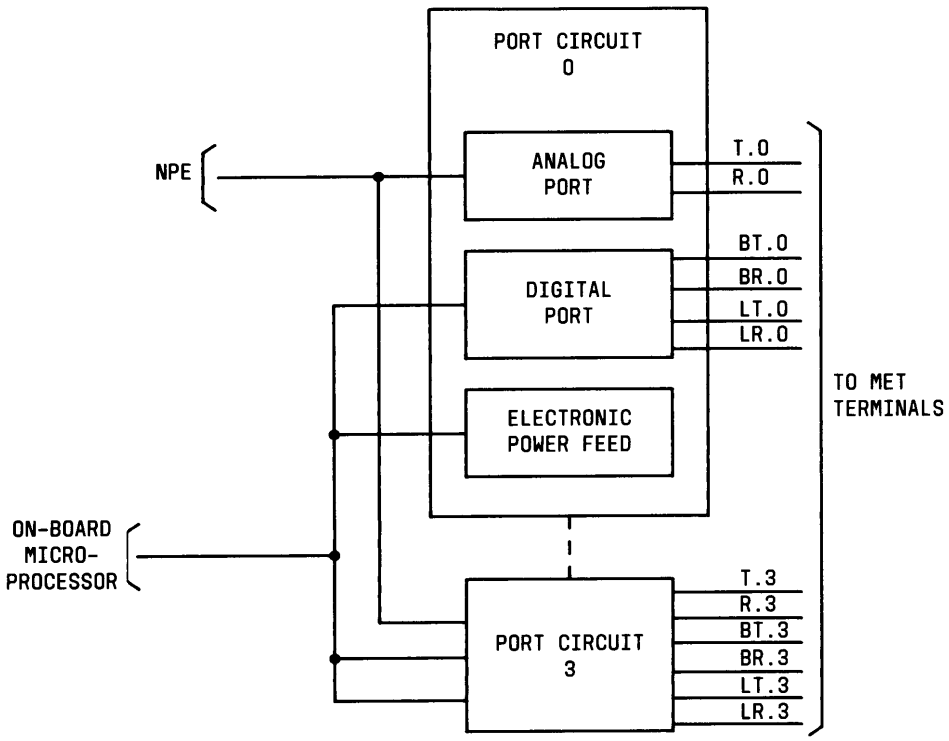


Figure 3-14. MET Line (TN735) Unique Circuitry

- Port Circuits

The four port circuits are identical. Each port circuit consists of an analog port, a digital port, and an electronic power feed device. The analog port circuit consists of a codec, a hybrid circuit, an electronic battery feed, and a power filter. The codec, hybrid circuit, and power filter perform the same function as in the Analog Line Circuit Pack (TN742). The electronic battery feed provides talking battery to the MET set, produces a controlled dc battery feed current for short and long loops, and detects when a MET set user lifts a receiver. The digital port circuit provides a full duplex channel over two 2-wire pairs. All outgoing lamp (LT, LR) and incoming button depression (BT, BR) information is carried on these channels. Ringing and

switchhook information is also sent over these channels.

The electronic power feed device provides phantomed -48 volt dc power for the MET terminals over the data channels. The electronic power feed device is a “smart” circuit breaker. When it senses an overcurrent condition, it indicates the condition on an output lead and goes into thermal shutdown if not turned off by the on-board microprocessor. When the overcurrent condition disappears, the circuit breaker can be turned on by the on-board microprocessor.

STARLAN Interface (ZTN84)

The STARLAN Interface (ZTN84) Circuit Pack functions as either a gateway or a bridge between System 25 and the AT&T STARLAN NETWORK (Release 2 of STARLAN only). The ZTN84 CP contains much of the circuitry common to the other CPs in the system, that is a Sanity and Control Interface (SAKI), a Network Processing Element (NPE), and a 8031 microprocessor. The CP also contains the circuitry required to perform the protocol conversion on the data as it travels from one system to the other. These devices include a 80186 microprocessor, 82586 coprocessor, four Octal Asynchronous Terminal Mode 2 to EIA Asynchronous LSI (OATMEAL) devices, and a logic sequencer. The 80186, the 82586, and the logic sequencer (PLS105N) work together to add and delete the protocol used by the Local Area Network (LAN), while the 80186 and the OATMEALS work together to add and delete the protocol used by the PBX.

The ZTN84 can support up to four circuit switch connections between the Private Branch Exchange (PBX) and the Local Area Network (LAN); this capability is provided by the four OATMEALS and the NPE, the latter being a four channel device. In providing a connection between the PBX and the LAN, capabilities such as file sharing, printer services, connections to hosts, and modem pooling may be accessible across systems.

The OATMEAL devices on the ZTN84 are used in such a way as to support asynchronous data communication at any of the standard rates ranging from 300 bps to 19.2 Kbps. The asynchronous protocol that is used is a subset of Digital Communications Protocol (DCP) Mode 2, as only “l” channel information is transmitted, where the data is formatted in High-Level Data Link Control (HDLC) frames.

The ZTN84 has been designed with a hardware interface that allows the CP to be connected to a STARLAN NETWORK as an OUT connection. This can be connected to a STARLAN NETWORK Extension Unit (NEU) IN connection, in a star configuration.

The design of the ZTN84 is not fully compatible with the daisy-chain arrangement of the STARLAN NETWORK, since much of the daisy-chain circuitry was left off of the card. For testing purposes, the card can be used in a limited daisy-chain arrangement, where the ZTN84 is connected to a personal computer (PC) that possesses an Network Access Unit (NAU). The ZTN84 and the PC should be the only two devices forming the LAN. The daisy-chain circuitry was omitted in order to reduce cost and save board space. It is also the architectural design of the system that the PBX be connected to the LAN by a NEU. The NEU can either be local, in the telephone room with the switch, or in a remote office.

Tie Trunk (TN760B)

The Tie Trunk Circuit Pack (Figure 3-1 5) interfaces four 6-wire tie trunks and the TDM bus. Two tip and ring pairs form a 4-wire analog transmission line. An E and M pair are used for signaling. The T and R pair transmit analog signals from the circuit pack. The T1 and R1 pair receive analog signals from the tie trunk. The E and M pair are dc signaling leads used for call setup handshaking. The E lead receives signals from the tie trunk and the M lead provides signals from the circuit pack. The TN760Bs four port circuits support Type 1, Type I Compatible, or Type V signaling. Incoming and outgoing trunks can be either automatic, immediate start, wink start, or delay dial. The Tie Trunk has the following unique circuitry:

- Ground Detector Circuit

  This circuit determines if a ground has been applied to the E lead. Ground detector inputs come from the port circuits as an analog current to the -48 volt dc supply. Its output is a port control point to the port I/O circuit.

- Port I/O Circuit

  This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the microprocessor.

- Port Circuits

  The four port circuits are identical, except for port 3 where part of the E-lead maintenance circuit is located. Each port circuit consists of a codec with associated input and output line transformers, analog operational amplifiers, a power filter, loop-around transistors, port control comparators, a relay driver, an electronic power feed device, an E-lead test maintenance circuit, and surge protection circuits.

  The codec converts the incoming 4-wire analog signal from the tie trunk to a PCM data signal. The codec converts the incoming PCM data signal from the NPE to an analog signal. Outgoing and incoming line transformers provide dc isolation to the tip and ring leads. Analog operational amplifiers provide amplification and buffering for the codec and network and loop-around gain compensation. Filtered power is provided to the codec and amplifiers.

  The loop-around transistors are under control of the port control comparators and provide a loop-around path for the signal for testing purposes. The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input operates the appropriate relay. The relays and electronic power feed device control the M-lead circuitry to provide the proper signaling handshake for call progress tones and dial pulse dialing.

  The electronic feed device provides a -48 volt dc current to the M-lead circuits. It also tests the M-lead circuits for opens or shorts and prevents uncontrolled operation during power-up. The E-lead test circuit provides a ground to the ground detector circuit for testing purposes. The surge protection circuitry provides lightning surge and power cross protection for the circuit pack. For each port circuit, E&M/Simplex and surge protection are selected by switch settings as shown on Figure 3-16.

The signaling type is administrable for each port. Table 3-B summarizes the conditions present as the transmit and receive control signals for each signaling type. Table 3-C lists the preferred TN760B tie trunk signaling format to be used in the likely-to-be-encountered installation situations.

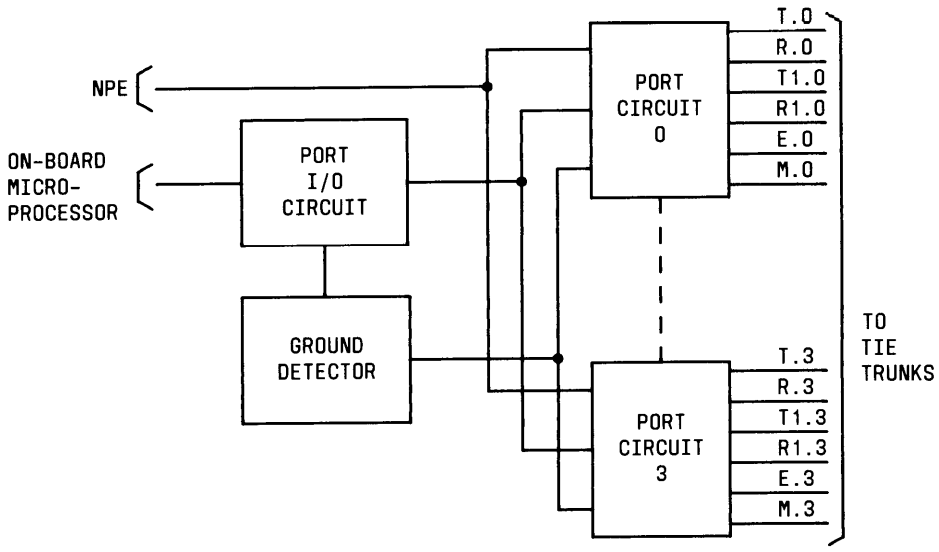


Figure 3-15. Tie Trunk (TN760B) Unique Circuitry

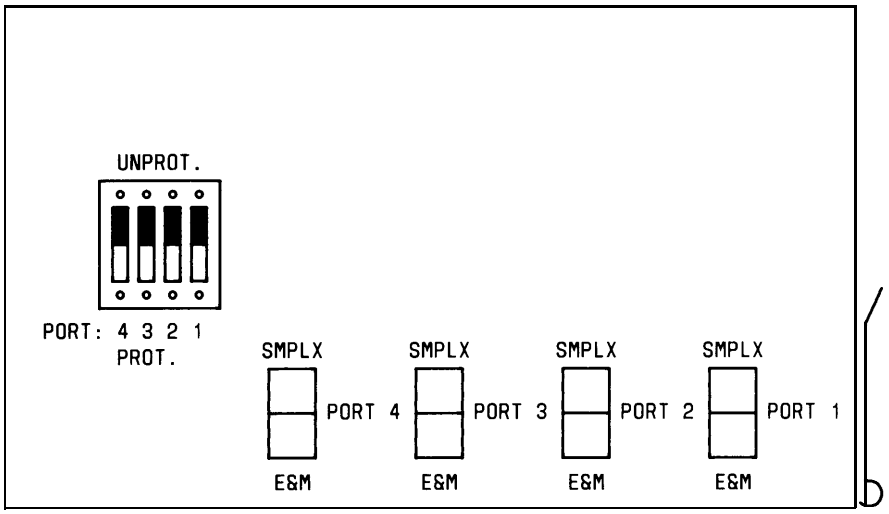


Figure 3-16. Tie Trunk (TN760B) Circuit Pack Option Switches

Table 3-B. Signaling Type Summary

Signaling Type	Transmit		Receive	
	On-Hook	Off-Hook	On-Hook	Off-Hook
I Std.	grd	bat	open/bat (*)	grd
I Compat.	open/bat (*)	grd	grd	open/bat (*)
V	open	grd	open	grd

* An open circuit is preferred over voltage.

Table 3-C. TN760B Tie Trunk Preferred Signaling Formats

Installation Situation			Preferred Signaling Format					
			System 25			Far End		
From	Circum-stances	To	Simplex or E&M (Note 1)	Signal Type (Note 2)	Protected or Unprotected (Note 1)	Simplex or E&M	Signal Type	Protected or Unprotected
S25	Co-Located	S25/S75	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Inter-Building	S25/S75	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Co-Located	S85	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Inter-Building	S85	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Co-Located	Dim.	E&M	Type I Compatible	Unprotected	E&M	Type I standard	Unprotected
S25	Inter-Building	Dim.	E&M	Type I Compatible	Protected	E&M	Type I Standard	Protected
S25	Co-Located	Other	E&M	Type I Compatible	Unprotected	E&M	Type I standard	Unprotected
S25	Inter-Building	Other	E&M	Type I Compatible	Protected	E&M	Type I Standard	(Note 3)
S25		Network Interface	E&M	Type I Standard	Unprotected	(Don't Care)	(Don't Care)	(Don't Care)

Notes: 1. Set by switches on Tie Trunk CP (Figure 3-1 6).
2. Set by System Administration of Port Options (Action 37).
3. Requires a protection unit.

Tip Ring Line (ZTN78)

The Tip Ring Line Circuit Pack interfaces eight analog tip and ring voice terminal lines (single-line voice terminals) and the TDM bus. Figure 3-17 shows the Tip and Ring Line unique circuitry. The TN742 can be used instead of the ZTN78 Tip Ring CP. The TN742 supports up to five bridged single-line voice terminals; however, only two can be off-hook at one time. The ZTN78 does not support bridged terminals. In addition, the TN742 supports out-of-building, extended, and off-premises stations, while the ZTN78 does not. The ZTN78 supports only a 1.2 Ringer Equivalency Number (REN).

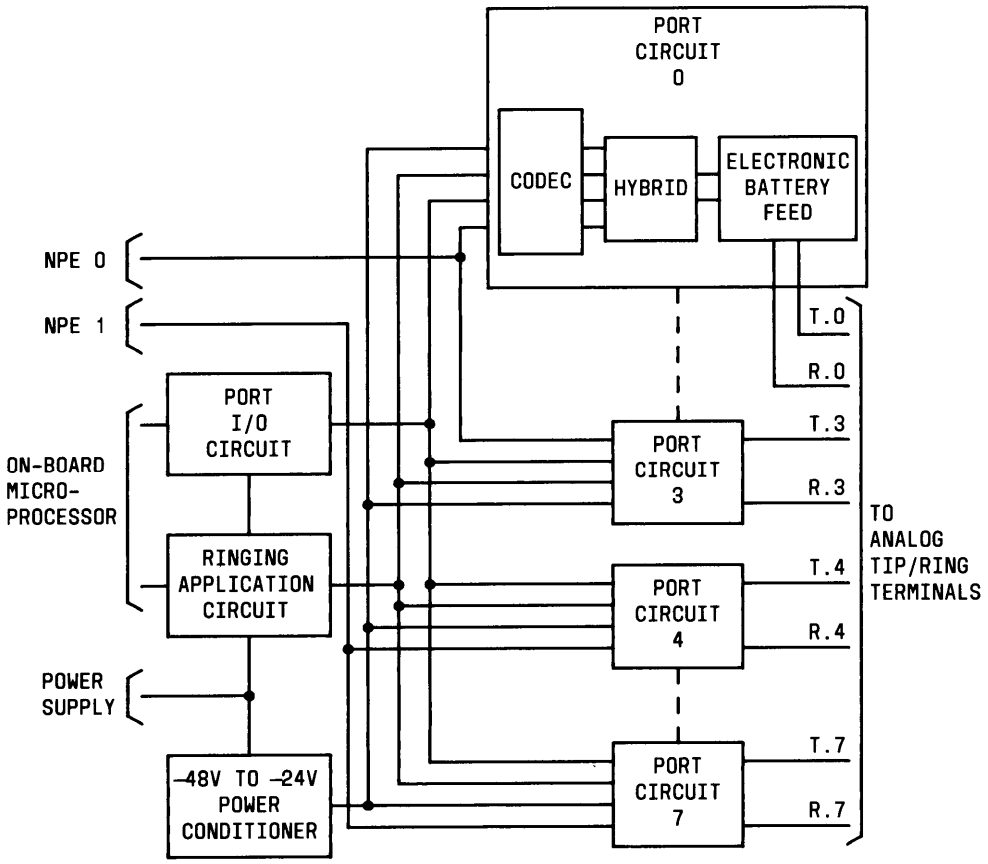


Figure 3-17. Tip Ring Line (ZTN78) Unique Circuitry

- Ringing Application Circuit

  This circuit receives ringing voltage from the power supply. It monitors ringing voltage and current and generates signals to the on-board microprocessor indicating zero ringing voltage and current. It also detects when a terminal user has lifted the receiver during ringing preventing the application of ringing to the terminal's handset receiver.

- Port I/O Circuit

  This circuit includes bus expanders connecting the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

- -48 V To -24 V Power Conditioner

  This circuit converts -48 V power from the power supply into a conditioned source of -24 V power for the electronic battery feed circuits.

- Eight Port Circuits

  Each port circuit is identical. A port circuit consists of a coder/decoder (codec), hybrid circuit, battery feed circuit, and ring relay.

  The codec is a 4-wire circuit that converts the NPEs output to an analog signal. Likewise, it converts the analog signal from a central office trunk to a PCM data signal to the NPE. The hybrid circuit converts the codec 4-wire analog signal to a 2-wire analog signal that is connected to the central office trunk by the line transformer.

  The battery feed circuit provides talking battery to the voice terminal. It also detects when a receiver is lifted, and provides the message waiting signal by periodically reducing the feed voltage to zero.

  The ring relay provides the interface between the ringing application circuit and the port circuit. It causes ringing to turn on and off.

System Resources

The System Resource Circuit Packs (CPs) are the Service Circuit (ZTN85), the Tone Detector (TN748B), and the Pooled Modem (TN758).

Service Circuit (ZTN85)

The Service Circuit CP (Figure 3-18) provides the clock signals of the system and generates and receives tones. It provides four touch-tone receivers, generates all tones for the system, and supplies the system clocks. The ZTN85 can support up to 75 Dual Tone Muitifrequency (DTMF) dialers depending on call traffic; the TN748Bs might be required in heavy traffic situations, even with less than 75 DTMF dialers. Each System 25 must contain one Service Circuit CP. Power for the circuit pack (+5 volts dc) is provided on the backplane. The Service Circuit has the following unique circuitry:

- Bus Buffers

  There are four bus buffers on the circuit pack. The clock driver and receive buffers interface three system clock signals (2.048 MHz, 8 kHz, and 160 kHz) to the TDM bus. Two buffers interface the system tones (see Table 3-A) between the TDM bus and the Service Circuit CP. Music is not provided by the Service Circuit but can be provided via a port interface on a Tip Ring Line CP (ZTN78).

- SAKI

  This circuit functions the same as in the SAKI in the common circuitry for the intelligent port circuits.

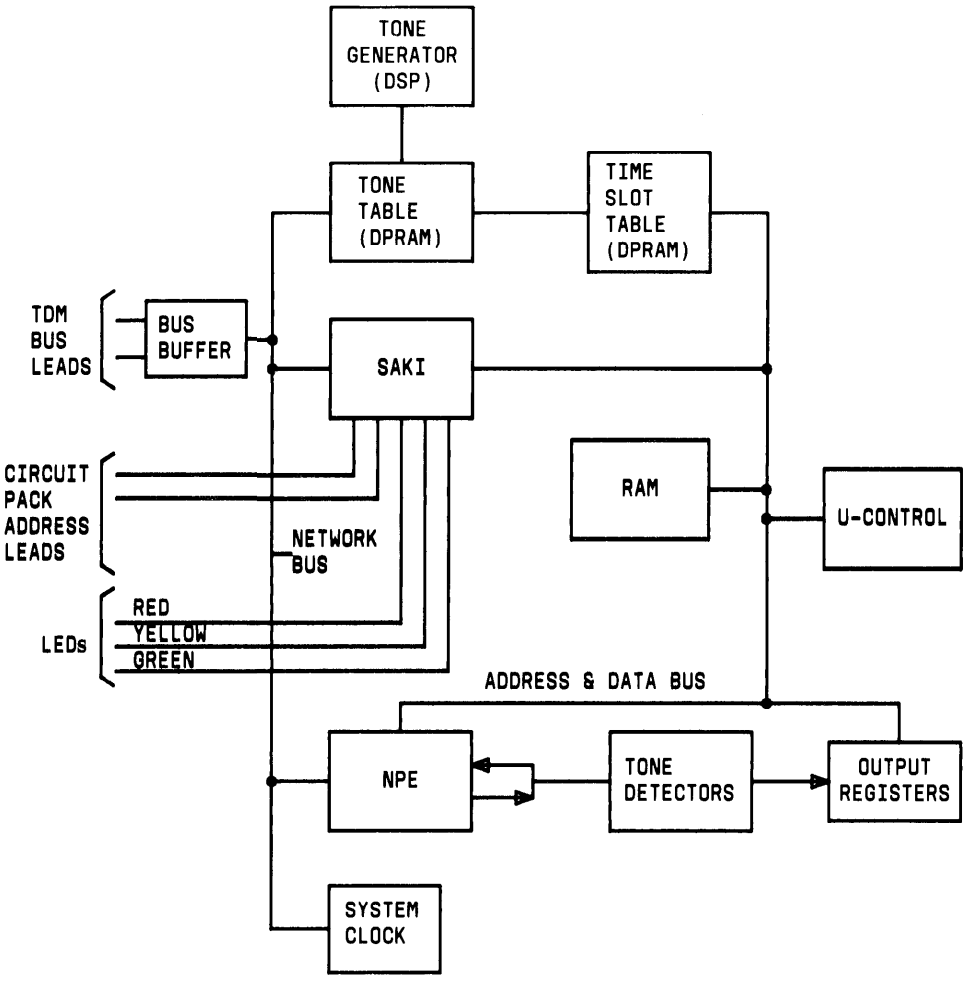


Figure 3-18. Service Circuit (ZTN85)

- On-Board Microprocessor With External RAM

  This circuit functions the same as the microprocessor in the common circuitry for the intelligent port circuits. In addition, it tells the dual-port RAM in the time slot table circuit the appropriate time slots in which to place a tone. The external RAM also has work space for complex tones (i.e., those tones that vary with time).

- Clock Circuit

  The clock circuit consists of a 20.48-MHz oscillator, various dividers, and shift registers. The clock circuit runs independently from the rest of the Service Circuit circuitry. The clock circuits start running when the circuit pack is first powered up and is not controlled by the on-board microprocessor.

  The output of the 20.48-MHz oscillator is fed to the clock divider. The divider divides by 10, 2560, and 128. These circuits produce the 2.048-MHz, 8-kHz, and 160-kHz clock signals, respectively. The clock generator feeds these signals to the clock driver/receiver bus buffer and the tone clock. The tone clock uses these signals to synchronize the counters in the tone generator and time slot table circuits with the TDM bus.

- Tone Generator

  The tone generator consists of a digital signal processor (DSP), a counter, and a dual-port tone RAM. The DSP operates at 10 MHz and produces 24 different tones. The dual-port tone RAM stores these tones in 24 different addresses. The counter under control of the tone clock causes the DSP to transmit one sample of each tone every 8-kHz. The counter is synchronized to the TDM bus and is offset to provide delay needed for access time.

- Time Slot Table and Counter

  The time slot table consists of a dual-port time slot table RAM and a counter. The dual-port RAM (DPRAM) contains 256 different addresses. These addresses correspond to the time slots on the TDM bus. The counter sequences through the time slot table addresses in the dual-port RAM and causes the proper tone(s) to be output by the dual-port tone RAM on TDM bus time slots.

- Tone Detector Ports

  The Service Circuit CP provides four Dual Tone Multifrequency (DTMF) detector port circuit interfaces via the TDM bus. Each port circuit is connected to an NPE serial input and output. Ports 0, 1, 2, and 3 are DTMF tone detectors with NPE loop-around paths.

  The four port circuits contain a DSP, NPE to DSP interface circuitry, a DSP restart circuit, and an interrupt generator. One DSP implements two tone receivers.

  The TDM bus signals are connected to the DSP in serial form from the NPEs by the DSP interface circuit. The DSP controls the output clocking of the NPE. The system framing signal is synchronized and connects to the DSP.

- Port I/O and Sanity Check Circuit

  This circuit interfaces the on-board microprocessor to the port circuits and checks the sanity status of the DSPs of the port circuit.

Tone Detector (TN748B)

The Tone Detector Circuit Pack provides four touch-tone receivers and two general purpose tone receivers that detect appropriate system and network tones on the TDM bus. The Tone Detector CP consists of the same common circuitry as the intelligent port circuits plus the unique circuits shown in Figure 3-19. The system can have a maximum of two Tone Detector CPs.

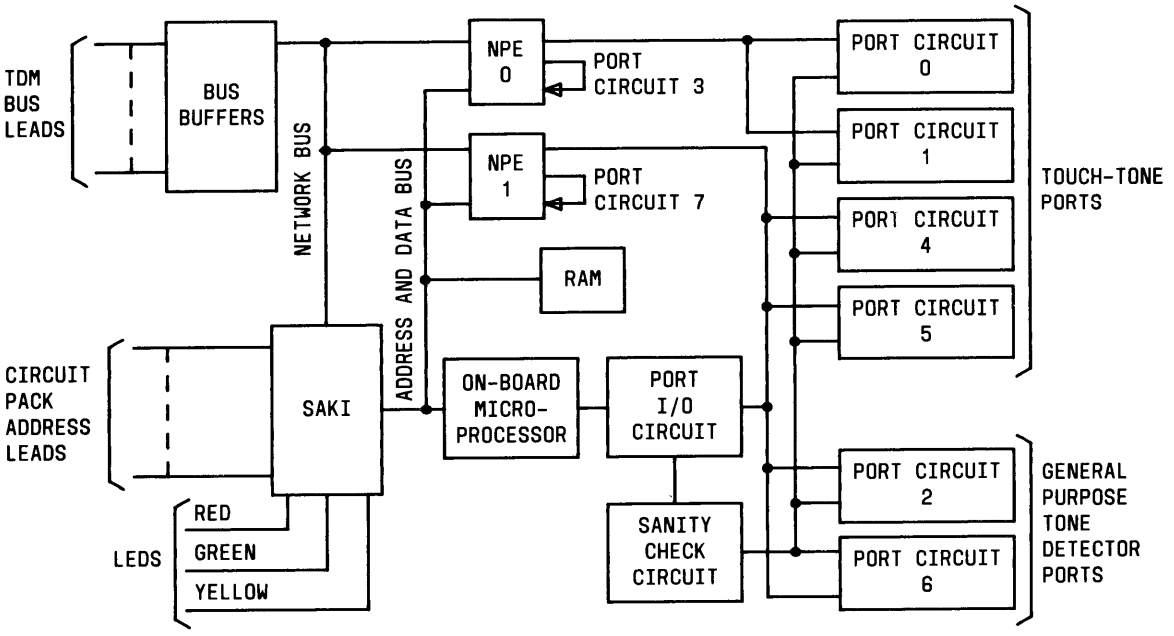


Figure 3-19. Tone Detector (TN748B)

- Port I/O and Sanity Check Circuit

  This circuit interfaces the on-board microprocessor to the port circuits and checks the sanity status of the port circuits Digital-Signal Processors (DSPs).

- Port Circuits

  There are eight port circuits. Six port circuits are connected to Network Processing Elements (NPEs). Port circuits 0, 1, 4, and 5 are DTMF tone detector ports. Each of the six port circuits has an associated Digital Signal Processor (DSP), NPE to DSP interface circuitry, a DSP restart circuit and an interrupt filter. Port circuits 2 and 6 are general purpose tone detector ports. Port circuits 3 and 7 provide digital loop-back testing of each NPE on the circuit pack.

  The NPE serializes TDM bus signals that are connected to the DSP in serial form from the NPEs by the DSP interface circuit. Serial clock and data signals connect directly from the NPE to the DSP. The system framing signal is synchronized and connects to the DSP.

  The DSP restart circuit controls the DSPs. When the on-board microprocessor is not functioning properly, the DSP restart circuit takes all of the DSPs out of service. It restarts each individual DSP under control of the port I/O and sanity check circuit.

  The touch-tone DSPs, under control of the on-board microprocessor, write data synchronously to the NPEs. The interrupt filter detects valid touch-tone signals and allows end-to-end transmission while blocking end-to-end touch-tone signaling.

Pooled Modem (TN758)

The Pooled Modem Circuit Pack supports 0-300 and 1200 bits per second (bps) data speeds and provides the following:

- Circuitry to provide a signal compatible with the modulation formats of the 212-series modems
- Modem emulation (see below)

Capability	Data Module Mode
0-300 Asynchronous	Low
300 Asynchronous	300 Asynchronous
1200 Asynchronous	1200 Asynchronous

- Modem control functions corresponding to 212A-series modem operations.

A maximum of two Pooled Modem CPs are allowed in a single cabinet (six in a 3-cabinet system).

The Pooled Modem CP (Figure 3-20) consists of common circuitry and two conversion resources. The conversion resource (port) allows communications between two dissimilar endpoints. For example, the Pooled Modem CP enables a digital data endpoint linked to an ADU connected to a port on the Data Line CP (TN726) to communicate with either a local analog data endpoint, such as a personal computer with a modem, or a remote host via a CO trunk connection. Each port has two connections to the TDM bus: one to the digital data endpoint via an ADU data module, and the other to an analog endpoint.

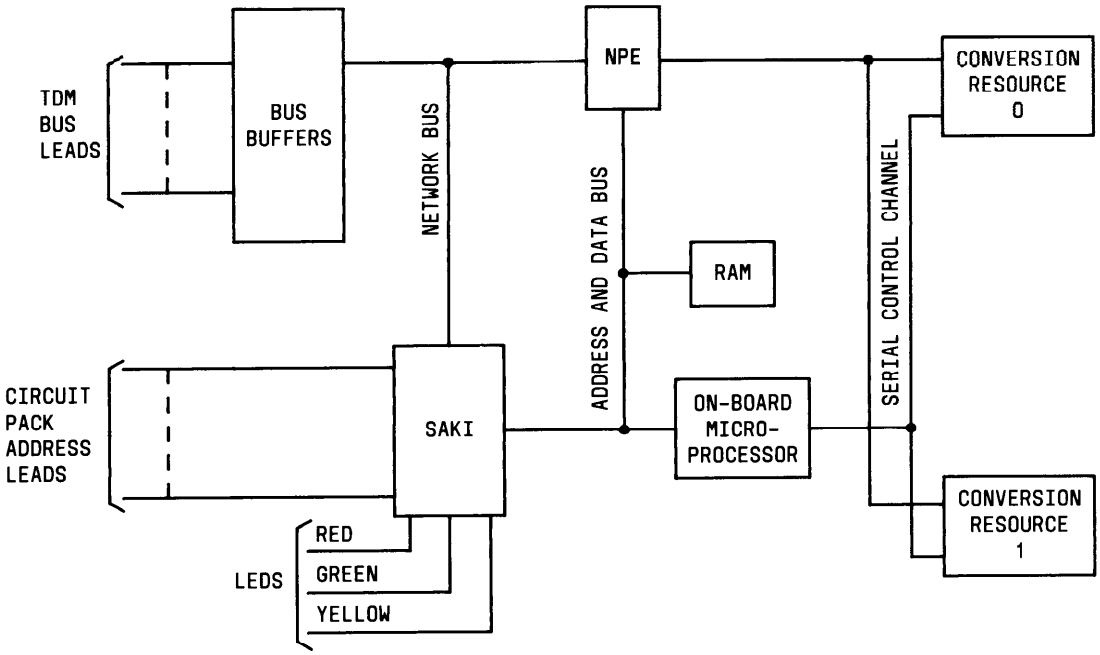


Figure 3-20. Pooled Modem (TN758)

- Common Circuitry

  The Pooled Modem contains the same common circuitry as port CPs.

- Conversion Resources

  The two conversion resources (port circuits) are identical and each contain the following:

  — Microprocessor

  — Transmit and Receive l-channel Controller (TRIC)

  — Universal Synchronous/Asynchronous Receiver and Transmitter (USART)

  — Data USART Clock (DUCK)

  — Digital Signal Processor (DSP).

  The microprocessor controls an on-board data module and modem. This microprocessor communicates with the port circuit microprocessor over a serial control channel. This channel allows the on-board microprocessor to send messages to the port circuit microprocessor specifying call startup information, option settings, information requests, various test modes, and call termination

information. It also allows the port circuit microprocessor to inform the on-board microprocessor of various port circuit status information.

The DUCK and TRIC interface l-channel information between the port circuit and the remote data module. The microprocessor controls the operation of the DUCK and the TRIC by programming their internal registers. The DUCK and TRIC together recreate the clock and serial data stream from the remote data module, and process an on-board clock and serial data stream for delivery to the remote data module. Control information, handshaking, and RS-232 control leads is passed between the port circuit microprocessor and the remote data module by the TRIC.

The USART interfaces the serial data stream of the DUCK to the conversion microprocessor. The USART can be programmed by the microprocessor to operate synchronously or asynchronously. The USART also does the following tasks for the port circuit microprocessor:

— Appends start and stop bits to parallel data received from the microprocessor in the asynchronous mode

— Converts serial data received from the DUCK to parallel data

— Buffers data in both directions

— Detects and generates break characters.

The DSP provides modem emulation. It interfaces the port circuit signal and the remote modem, The microprocessor directs the DSP to execute one of many programs. The DSP produces data, carrier detection, and timing information for the port circuit microprocessor.

Software

The System software consists of switched services, administrative, and maintenance software. This software runs on top of the real-time operating system software.

Switched Services Software

The switched services software provides voice and data call processing. This software resides in the Common Control circuitry and in the 8-bit on-board microprocessors located in the port and service circuits.

The switched services software uses the operating system to provide a process based, message passing, execution environment. The operating system scheduler provides scheduling for the software according to process priority.

Administrative Software

The administrative software provides the control for system rearrangement and change via the System Administration Terminal (SAT). This software resides in the CPU/MEM Circuit Pack and does the following functions:

- Organizes the translation data for administrable entities in the system in a form that can be viewed and changed at the System Administration Terminal.
- Tests entered data for consistency with data previously entered in order to avoid such errors as the assignment of the same extension number to two voice terminals. An erroneous or inconsistent data entry is disallowed and an error message is provided.
- Causes the translation data to be downloaded, on command, to an optional Digital Tape Unit (DTU).

Maintenance Software

The maintenance software provides automatic periodic testing of maintenance objects within the system as well as consistency tests among the call status tables within the system. In addition, demand testing is initiated when the system detects a condition requiring a need for testing. Software tables are provided for storing error records. The records can be accessed by maintenance personnel via the SAT. A Permanent System Alarm (a serious error) causes an alarm indicator on the attendant console to light and an error record to be stored in the error table.

Memory Allocation

The system software, like the hardware, is identified by release and version number. Each version identifies a particular memory configuration for the release number. Main memory is located in the Common Control circuitry, that is, the CPU/MEM Circuit Pack.

Real-Time Constraints

Real-time constraints are a function of the speed of the common control circuitry and the traffic load. The switch is designed so that many time-consuming and repetitious functions are performed by processors in the port and service circuit packs, thus relieving the common control circuits.

Traffic load, defined as the sum of static and dynamic loads, is a function of the number of features that are executed, the frequency with which they are executed, the system configuration, and the instantaneous (peak) call processing load. The configuration contribution to load is known as dynamic load. The static load consists of maintenance and audit routines.

Software Partitioning

As shown in Figure 3-21, System 25 software is comprised of various modules, each supporting a particular process. Typical modules (referred to as tasks) include the following:

- Administration
- Station Call Processing
- Station Message Detail Recording (SMDR) Call Record Processing
- Trunk Call Processing
- Dial Plan Manager
- Event Timer
- Save/Restore (Administration function)
- Maintenance and Audit Functions.

Specific software tasks are associated with the memory and call processing portions of the CPU/MEM, the TDM Bus, and the Port Circuits.

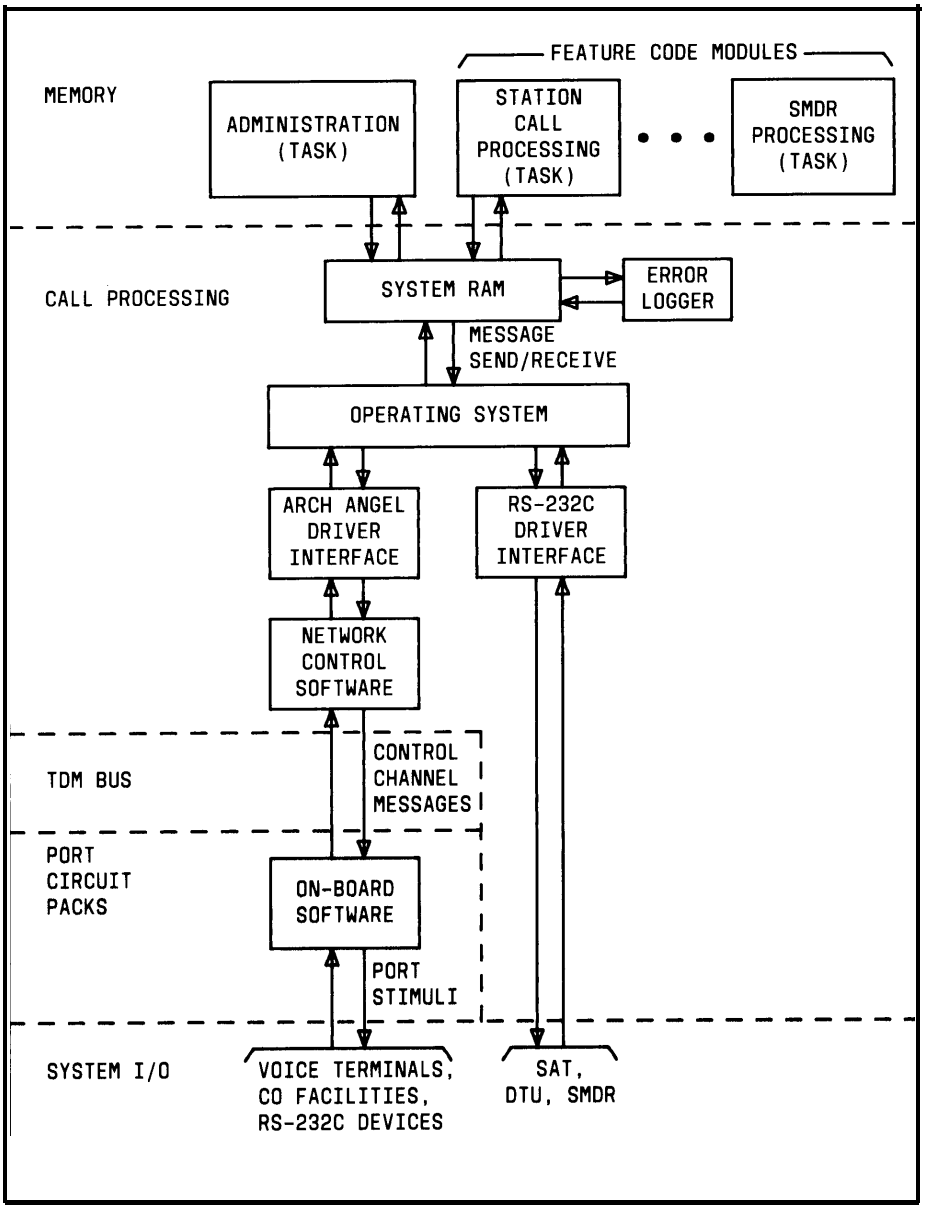


Figure 3-21. System Software Partitioning

Memory

Administration and Feature Code Modules, which includes Station Call Processing, are software tasks associated with memory. Each task controls the storage and movement of data and messages between elements in the system.

- Administration

  Provides for administration of station and system features. This software also supports maintenance procedures related to error checking and diagnosing trouble.

- Feature Code Modules

  Includes the software that receives and sends data to/from the Operating System, as well as control of all voice and data features supported by the system. Station Call Processing includes the processing of messages and data associated with voice terminal on-hook/off-hook indications, associated port identifications, button and LED operations, etc. The SMDR software generates SMDR records associated with a particular call. The records are then sent to the System RAM for storage and then to the SMDR output channel.

Call Processing

The following circuits support software tasks.

- System RAM provides for the storage of the following:
  - — Variables for the various software tasks
  - — System translations
  - — Error Records
  - — Feature Code Data
  - — Stack.
- Error Logger

  Prioritizes and stores system errors; lights the Alarm LED on the Attendant Console when a serious error is detected. The errors are stored in three error records in System RAM:
  - — Permanent System Alarms
  - — Transient System Errors
  - — Most Recent System Errors.

- Operating System (OS)

  Controls all message and data flow to/from memory, to the Arch Angel Driver Interface, to the microprocessors on the port circuit packs, and to the RS-232 driver interfaces. Messages destined for a particular task are queued until the associated task can receive them. When a task has completed a particular process, the next message is obtained from the message queue of the tasks. The OS provides an interval timer that is used to time tasks. Processes that exceed the set interval (about 60 seconds) are terminated by the OS.

- Arch Angel Driver Interface

  Provides an interface between the OS and Network Control.

- RS-232 Driver Interface

  Handles the flow of information between the CPU/MEM CP and the peripheral equipment of the system (i.e., System Administration Terminal or Advanced Administration PC, Digital Tape Unit, and SMDR Output Device.)

TDM Bus

Provides an electronic link between the system port circuits (including System Resources) and between the CPU/MEM and port circuits.

Port Circuit Packs

Each port circuit pack has on-board software that provides for the sending/receiving of Network Control messages and data. Circuit pack status messages are also sent to the Network Control software.

Step-By-Step Call Description

The following is a description of a call originated between two multiline voice terminals.

1. A microprocessor on a station port circuit pack (port controller) continually monitors associated port circuits for switchhook status/change and button presses.
2. When a user goes off-hook, the port controller detects the change.
3. The port controller sends an off-hook up-link message along with port identification to the Call Processor Network Controller (CPNC) via the TDM bus.
4. The CPNC accepts the message and forwards it to the Operating System (OS) via the Arch Angel Driver Interface.

5. The OS checks a message directory to determine which task (i.e., software module) is to receive the message. A function of the OS referred to as the “transformer” determines it has a message for the Station Call Processing task and queues the message in RAM.

6. The Station Call Processing task retrieves its message and interprets it as a call origination. The task determines whether there is an idle call appearance button (System Access button) on the called voice terminal. If so, two available time slots are reserved for the connection.

7. The task sends downlink messages to the port circuit via the OS. The messages instruct the port circuit to listen for dial tone on a specified time slot and to light the call appearance status LED on the terminal.

8. When the user dials the first digit, the port circuit determines the digit dialed. It then listens to appropriate time slots on the TDM bus for the two tones used to generate an equivalent DTMF signal. It then removes dial tone and feeds the DTMF signal back to the user until the user releases the button.

9. The port circuit sends an up-link message with each digit dialed to the OS that routes them to the Dial Plan Manager (DPM).

10. The DPM collects the dialed digits and determines that the call is a station-to-station call.

11. When the DPM collects enough digits to identify an extension number it stops collecting digits.

    Note: If the extension number dialed is invalid, the DPM sends a down-link message to the port circuit instructing it to listen to time slot 07 (Reorder Tone) that is then heard by the user. Go to Step 18.

12. A down-link message is sent to the originating port instructing it to listen to time slot 06 (busy) or 08 (ringing), as appropriate. Go to Step 18 for Busy Tone or an unanswered call.

13. Station Call Processing sends a down-link message to the station port circuit pack associated with the called extension to turn on the ringer of the terminal, and to flash the call appearance LED.

14. When the called party lifts the receiver, the associated port circuit pack controller sends a off-hook message to the OS as before.

15. The Station Call Processing task, when it receives the message interprets the off-hook message as an answer.

16. The task sends a down-link message to the called port circuit to turn off the ringer and to change the flashing LED to steadily lighted.

17. Down-link messages are sent to the port circuits assigning talk and listen time slots for the connection.

18. When either of the parties hangs up, the associated port circuit controller sends an up-link message to the Station Call Processing task.

19. Station Call Processing interprets the on-hook message as the end of the call.

20. The task then sends a down-link message to the port circuit pack controllers to disconnect the time slot connections and turn off the LEDs associated with the calls.

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HARDWARE DESCRIPTION

This Section provides descriptions of System 25 hardware components and their functions. The hardware is covered under the following major headings:

- **System Cabinets:** Includes circuit pack (CP) carriers, power supplies, wiring, and cooling equipment.
- **Circuit Packs:** Includes detailed information on CPs.
- **Terminal Equipment:** Equipment that can be connected to voice or data station ports.
- **Peripheral Equipment:** Equipment that can be connected to the CPU/MEM CP.
- **Auxiliary Equipment:** Service- and feature-related supporting equipment.
- **Connectivity:** Equipment and arrangements for interconnecting the various elements of System 25 hardware.

Note: Equipment that is directly associated with a specific feature or service covered in Section 2 is also described there; such equipment is noted in this Hardware Description section, with a reference to the appropriate heading in Section 2.

All system hardware, except Cabinet 1 equipped with a CPU/MEM, Service Circuit, and associated cables, is optional.

A listing of Product Element Codes (PECs), Apparatus Codes, and Comcodes is provided in Section 7, “Parts Information.”

System Cabinets (J58901A1 L4)

The system can consist of one, two, or three cabinets (Figure 4-1). Each cabinet contains its own power supply and cooling system. A CP carrier frame is integrated into each cabinet. Depending on the circuit pack complement, the cabinet/circuit pack configuration is as follows:

- Cabinet 1 (always required)—Contains the CPU/MEM CP and the Service Circuit of the system and can also contain up to ten port CPs.
- Cabinet 2 or 3 (optional) —Provides mounting for up to 12 port CPs each.

The CPs receive power, control signals, and data via the backplane bus of the carrier and associated 25-pair connector interfaces. In multiple cabinet systems, the backplane buses are linked with a bus extender cable (J58901A4 L3).

The cabinets have a brown front cover with beige top and sides. The front cover has a system identification stripe across the top. The top has four indentations to facilitate the stacking of cabinets.

Each cabinet is constructed of sheet metal and is 13 inches high, 17 inches wide, and 21 inches deep and weighs about 75 pounds fully loaded. A 3-cabinet system occupies a vertical space of about 40 inches. It is recommended that the cabinets be placed on a desk- or table-top. They *must not* be placed on a floor where cleaning solutions and dirt can get into them. Refer to Section 6, "Environmental Requirements" for equipment area considerations and associated floor plan recommendations.

The front cover of the cabinet is secured by four screws. These screws must be loosened slightly before the cover can be removed. When the cover is removed, access is provided to the CPs, a replaceable air filter mounted just under the CP carrier frame, and two cooling fans. The cooling fans are mounted on an assembly that, when unscrewed, provides access to the power supply. Air intake is at the bottom of the cabinet and exhaust is vented at the left side of the front cover.

Each cabinet has its own power supply mounted to the left of the CP carrier. The power supply is 3 inches wide and weighs about 9 pounds. Voltage and current supplied to the carrier are: +5 V dc at 35A, -5 V dc at 3A, -48 V dc at 3A, and 90 V ac at 0.16A.

On the front of the supply is a green Light-Emitting Diode (LED) that, when lighted, indicates that the +5 V dc is available and within limits. The LED can be viewed through the slotted area on the front cover, and is just behind the fan located at the top left edge of the cabinet.

Mounted on the back of the cabinet (Figure 4-2) is the aluminum grounding block with four terminating positions, an ac input power receptacle, a power On/Off switch [(1)= ON, (0)=OFF] and twelve 25-pair connectors. The ground block is connected to dc ground on the carrier backplane at a location near the power supply. The 25-pair connectors provide an interface between cross-connect wiring and the CPs immediately behind each connector. Two slots are provided in the rear cover just above the 25-pair connectors for the Time Division Multiplex (TDM) bus extender cable. The TDM cable is used to connect 2- or 3-cabinet systems in a daisy-chain configuration and provides control and data signals between cabinets. The Cabinet 1 ground block is connected to the single-point ground of the system using 6 AWG wire. Separate 6 AWG wires are then connected from the Cabinet 1 ground block to Cabinets 2 and 3 ground blocks. The Cabinet 1 ground block is also connected to the Coupled Bonding Conductor. An information label is provided across the top portion of the rear panel on each cabinet. The label provides cabinet identification, input electrical requirements, caution and warning notes, and FCC, CSA, and UL labels.

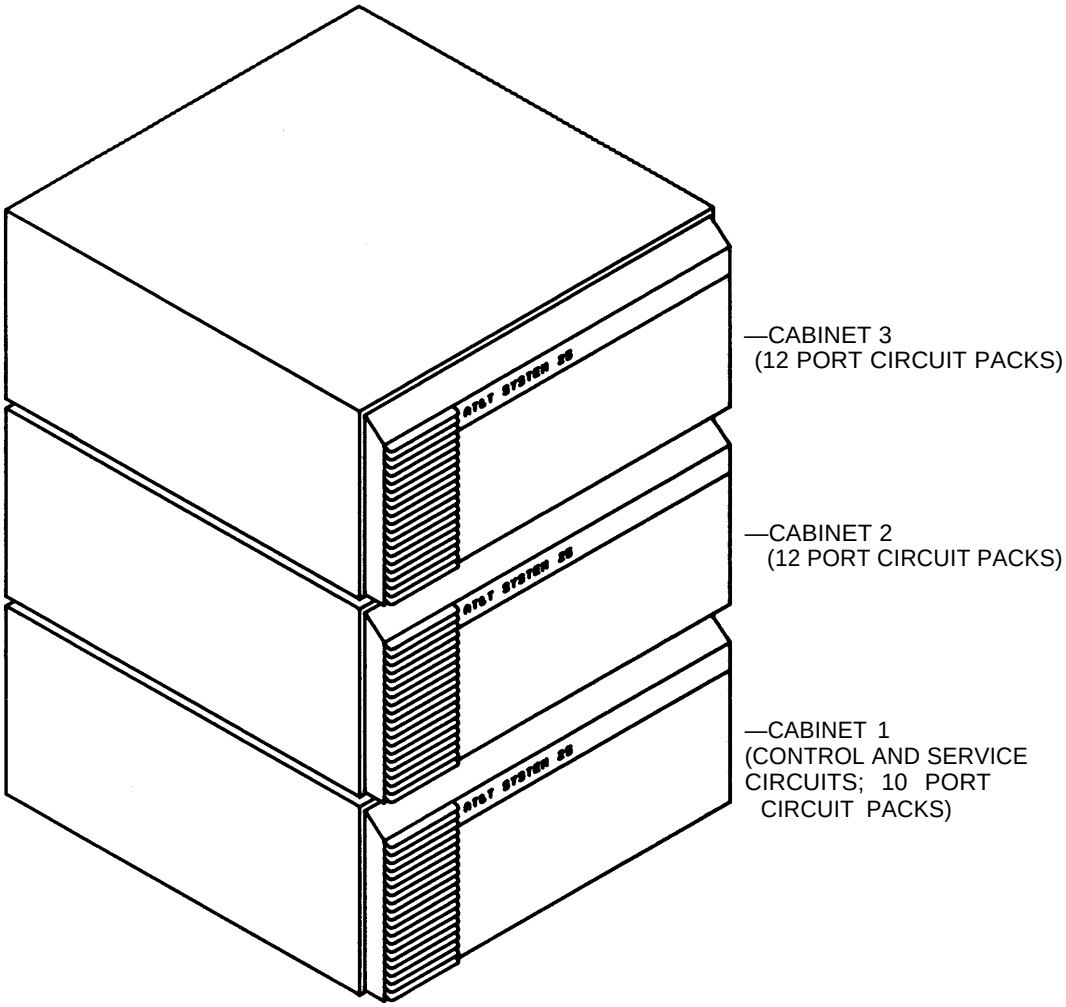


Figure 4-1. System 25 Cabinets (J58901A)—3-Cabinet System

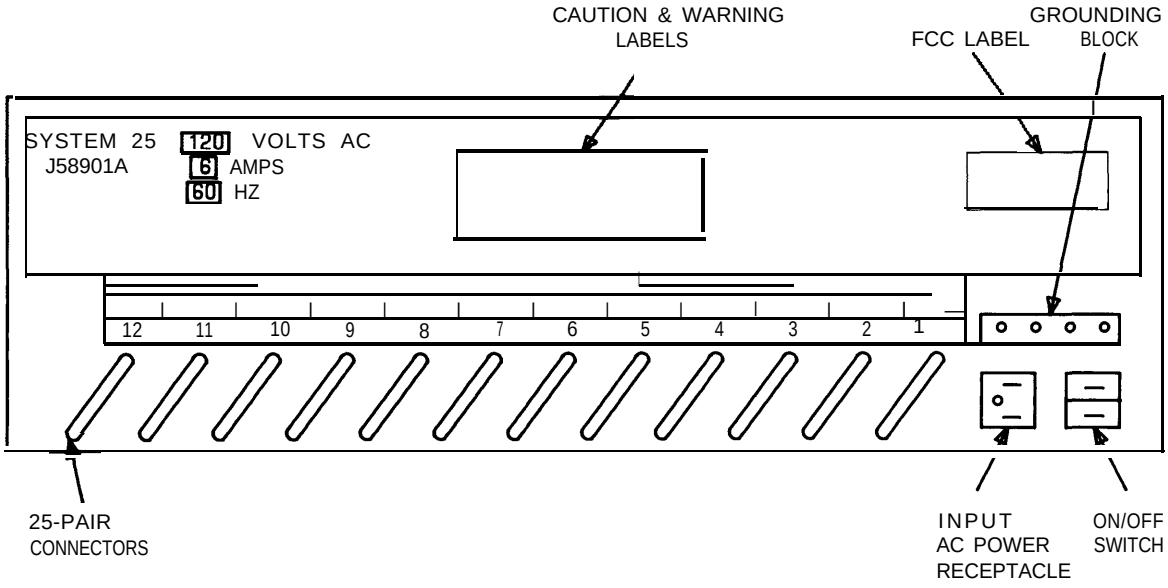


Figure 4-2. System Cabinet (J58901A)—Rear View

Cabinet 1 (Control and Port Circuits)

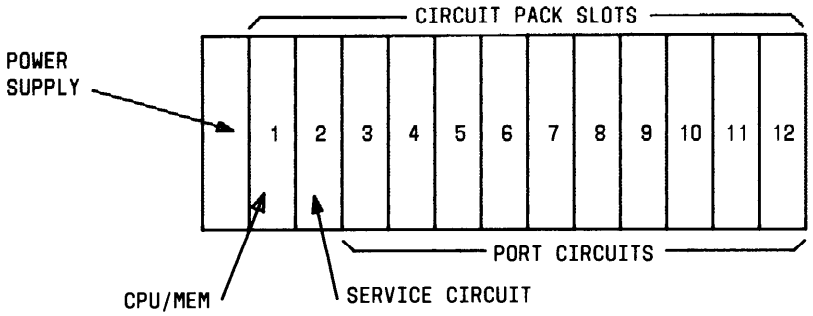
Cabinet 1 (Figure 4-3) is always required. It provides mounting space for 12 CPs and can support a small telecommunications system (for example, 50 to 60 stations and 10 to 15 trunks.)

Cabinet 1 contains the CPU/MEM CP and the Service Circuit, which must be mounted in CP slots 1 and 2, respectively. Slots 3 through 12 (ten total) provide mounting for the various port CPs that can be used. Any port CP can be mounted in any of these ten slots.

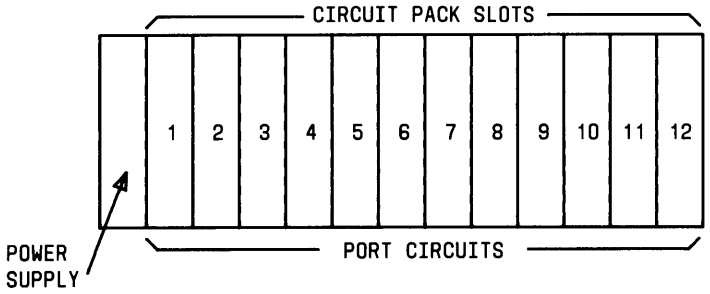
The Tone Detector and Pooled Modem CPs (referred to as System Resource CPs) can also be mounted in the port CP slots. Circuit packs are described in this Section under the heading “Circuit Packs”.

Cabinets 2 and 3 (Port Circuits)

Cabinet 2 and Cabinet 3 (Figure 4-3) can be added to provide mounting space for additional port CPs (12 maximum each) required for larger systems. The Tone Detector and Pooled Modem CPs can also be mounted in these cabinets. These cabinets are simply stacked on top of Cabinet 1. Table 4-A summarizes port CP capacity of 1-, 2-, or 3-cabinet systems.



a) CABINET 1 - (MOUNTING FOR CONTROL & PORT CIRCUIT PACKS)



(b) CABINET 2 OR 3 - (MOUNTING FOR PORT CIRCUIT PACKS)

NOTES:

1. REFER TO TECHNICAL SPECIFICATIONS, SECTION 5 FOR CIRCUIT PACK UNIT LOAD INFORMATION.
2. DIVIDE THE TOTAL NUMBER OF VOICE TERMINAL AND TRUNK CIRCUIT PACKS BETWEEN THE CABINETS USED.
3. MOUNT VOICE TERMINAL CIRCUIT PACKS FROM THE RIGHT, TRUNK CIRCUIT PACKS FROM THE LEFT.
4. COMMON CONTROL CIRCUIT PACK *
   - CPU/MEM (ZTN129)
5. SYSTEM RESOURCE CIRCUIT PACKS #
   - SERVICE CIRCUIT (ZTN85)
   - TONE DETECTOR (TN748B)
   - POOLED MODEM (TN758) MAXIMUM 2-PER CABINET)
6. PORT CIRCUIT PACKS †
   - TIP RING LINE (ZTN78)
   - ATL LINE (ZTN79)
   - MET LINE (TN735)
   - ANALOG LINE (TN742)
   - AUXILIARY TRUNK (TN763)
   - STARLAN INTERFACE (ZTN84)
   - DATA LINE (TN726)
   - GROUND START TRUNK (ZTN76)
   - LOOP START TRUNK (ZTN77)
   - DID TRUNK (TN753)
   - TIE TRUNK (TN760B)

* CIRCUIT PACK MUST BE MOUNTED IN SLOT 1 OF CABINET 1.

† UNIVERSAL PORT CIRCUIT PACKS CAN BE MOUNTED IN ANY AVAILABLE PORT SLOT

\# SERVICE CIRCUIT MUST BE MOUNTED IN SLOT 2 OF CABINET 1. POOLED MODEM AND TONE DETECTOR MAY BE MOUNTED IN ANY PORT CIRCUIT SLOT.

Figure 4-3. System Circuit Pack Configurations

Cabinet Address Plug

An address plug is provided on the middle of the backplane of each cabinet (accessible after removing the top rear cover) and is used to designate the cabinet number to the software. When address plug is plugged into the designated area at CP slot 5, the cabinet is identified as Cabinet 1; at slot 6 as Cabinet 2, and at slot 7 as Cabinet 3.

Table 4-A. Total Port Circuit Packs Per System

NUMBER OF CABINETS	CABINET NUMBER *	PORT CIRCUIT PACKS	TOTAL PORT CIRCUIT PACKS †
1	Cabinet 1	10	10
2	Cabinet 1	10	22
	Cabinet 2	12	
3	Cabinet 1	10	34
	Cabinet 2	12	
	Cabinet3	12	

* Cabinet 1 (always required)—Provides mounting for CPU/MEM, Service Circuit, and Port CPs including Tone Detectors and Pooled Modems.

Cabinet 2 and 3 (optional)—Provides mounting for Port CPs including Tone Detectors and Pooled Modems.

† The Number of Ports per CP is specified in the CP descriptions.

Circuit Packs

This part describes required and optional System 25 Circuit Packs (CPs) and their compatibility and features. Required CPs are the CPU/MEM and the Service Circuit. Optional CPs are the Station Port CPs, Trunk Port CPs, Pooled Modems, and Tone Detectors; the latter two CPs, plus the Service Circuit, are also classified as System Resources. Table 4-B lists the CPs of System 25. For more detailed functional descriptions of the CPs, refer to Section 3 of this manual.

Table 4-B. System Circuit Packs

CIRCUIT PACK	TITLE	CIRCUIT PACK TYPE *	NUMBER OF PORTS
TN726	Data Line	P	8
TN735	MET Line	P	4
TN742	Analog Line	P	8
TN748B	Tone Detector †	R	4
TN753	DID Trunk	P	8
TN758	Pooled Modem †	R	2
TN760B	Tie Trunk	P	4
TN763	Auxiliary Trunk	P	4
ZTN76	Ground Start Trunk	P	8
ZTN77	Loop Start Trunk	P	8
ZTN78	Tip Ring Line	P	8
ZTN79	ATL Line	P	8
ZTN129	CPU/MEM	C	4
ZTN84	STARLAN Interface	P	4
ZTN85	Service Circuit t	R	4

* P = Port, C = Control, R = System Resource.

† System Resource Circuits (Tone Detector, Pooled Modem, Service Circuit) ports are internal to the system. These ports are not connected to external equipment via 25-pair connectors.

Required Circuit Packs

The following CPs are provided with all systems and must be mounted in Cabinet 1:

- ZTN129 CPU/MEM (Call Processing Unit/Memory)

  The ZTN129 (one per system) provides a central processing unit, Random-Access Memory (RAM) (memory) for call and feature processing, interrupt controller, programmable timers, real time clock, status display, processor bus interface, and four interface ports. It also provides Read-Only Memory (ROM) for translations and a 1200-baud answer-only modem for Remote Initialization and Maintenance Service (RIMS) access. The four interface ports provide the following interfaces:

  — Port l- System Administration Terminal (SAT)

  — Port 2- Station Message Detail Recording (SMDR) equipment

  — Port 3- Digital Tape Unit

  — Port 4- Reserved for future use.

  The CPU/MEM also provides -48 V dc control on ports 7 and 8 for Emergency Transfer Units.

  The CPU/MEM CP must be mounted in slot 1 of Cabinet 1.

- ZTN85 Service Circuit

  The ZTN85 (one per system) provides four touch-tone receivers, generates all system tones, and supplies the system clocks. The ZTN85 can support up to 75 voice terminal stations.

  The Service Circuit CP must be mounted in slot 2 of Cabinet 1.

Optional Circuit Packs

The following CPs are optional and can be mounted in any slot not occupied by the required CPs.

Station Port Circuit Packs

- TN726 Data Line

  Provides eight ports for Asynchronous Data Units (ADUs). Used for in-building service within 2000 feet of the system cabinets. Data speeds from 300 bps to 19.2 Kbps are supported. Service beyond 2000 feet at less than 19.2 Kbps is supported; see Section 5 “Technical Specifications. ” Extends a serial communications link from the system to data equipment over standard station wiring.

- TN735 MET Line

  Provides four ports for Multibutton Electronic Telephone (MET) sets. Used for in-building service within 1000 feet of the system cabinets.

- TN742 Analog Line

  Provides eight ports for single-line voice terminals with or without non-neon message waiting lamps. Also supports Off Premises Stations (OPS) and out-of-building service. Auxiliary equipment interfaces are also supported. Used for service within 24,000 feet of the system cabinets. Five voice terminals can be bridged onto each port. Only two terminals can be off-hook simultaneously on each port; otherwise, transmission can be degraded.

  Note: The Off-Premises Stations must be FCC registered.

- ZTN78 Tip Ring Line

  Provides eight ports for single-line sets with or without non-neon message waiting lamps. Used for in-building nonbridged voice terminal service within 2000 feet of the system cabinets.

  Note: Equipment connected to the ZTN78 Tip Ring Line CP must meet the following requirements:

  — AC impedance: 600 ohms

  — DC current: Less than 30 ma at 48 volts

  — Ringer Equivalence Number (REN): Less than 1.15 (set plus adjuncts).

  The ZTN78 CP ports can also be used for interface with the AT&T VOICE POWER Voice Message System (VMS).

- ZTN79 ATL Line

  Provides eight ports for MERLIN® Communications System voice terminals. Used for service within 2000 feet of the system cabinets; terminals more than 1000 feet from the system cabinet require local power. Off-premises extensions are not supported. Out-of-Building stations require in-Range Out-of-Building (lROB) units.

- ZTN84 STARLAN Interface

  Provides four ports for interface with AT&T STARLAN NETWORKs (Release 2 of STARLAN only). It supports four connections between data endpoints connected to the PBX and data endpoints connected to the STARLAN NETWORK. The Network Extension Unit must be collocated with the System 25 cabinets.

Trunk Port Circuit Packs

- TN753 DID Trunk

  Provides eight ports for immediate-start or wink-start Direct Inward Dialing (DID) trunks.

- TN760B Tie Trunk

  Provides four ports for Type 1 E&M, Type 1 E&M Compatible, or Type 5 Simplex tie trunks. Operating protocols include automatic, immediate-start, wink-start, or delay dial. The TN760B contains option switches for supporting the following signaling formats:

  — Type 1 E&M Standard (Unprotected)

  — Type 1 E&M Compatible (Unprotected)

  — Type 1 E&M Compatible (Protected)

  — Type 5 Simplex

- TN763 Auxiliary Trunk

  Provides four ports for on-premises auxiliary equipment (paging systems and dictation systems).

- ZTN76 Ground Start Trunk

  Provides eight ports for Ground Start Central Office (CO), Foreign Exchange (FX), or Wide Area Telephone Service (WATS) trunks.

- ZTN77 Loop Start Trunk

  Provides eight ports for loop-start CO, FX, or WATS trunks.

Refer to Section 9, “Glossary” for Ground Start and Loop Start definitions. Ground Start trunks are recommended for use where possible.

Trunk specifications are provided in Section 5, “Technical Specifications.”

System Resource Circuit Packs

- TN748B Tone Detector

  Provides four touch-tone receivers; the system can have a *maximum* of two TN748Bs, depending on the number of stations and the conditions in Table 4-C.

  One TN748B CP must be used in addition to the ZTN85 Service Circuit if the system has from 75 up to 150 total stations or if required in Table 4-C.

  Two TN748Bs must be used in addition to the ZTN85 Service Circuit if the system has more than 150 total stations or if required in Table 4-C.

Table 4-C. Tone Detector Requirements

Traffic (Calls/Hr.)	Number of Tone Detector (TN748B) CPs		
	No Account Codes Used and no Voice Message System (VMS)	Automated Attendant or VMS	Automated Attendant and VMS
110	0	0	1
180	0	1	1
350	o	1	2
420	1	1	2
610	1	2	2
710	1	2	Unsupported
1100	1	Unsupported	Unsupported
1400	2	Unsupported	Unsupported

- TN758 Pooled Modem

  Provides two integrated 212-modem compatible conversion resources for switched connections between analog endpoints (modems), or a digital endpoint and an analog endpoint. A maximum of two TN758s (four conversion resources) is permitted in each cabinet.

Circuit Pack Compatibility

The following System 75 CPs can be used in System 25, if required:

- The TN742 Analog Line can be used instead of the ZTN78 Tip Ring CP. The TN742 supports bridged stations and out-of-building or Off-Premises Stations (OPS), the ZTN78 does not.
- The TN762B Hybrid Line (Version 4 or later) can be used instead of the ZTN79.
- The TN747 CO Trunk can be used instead of the ZTN76 (Ground Start Trunk) or the ZTN77 (Loop Start Trunk).

Circuit Pack Features

All system CPs have the following features:

- Solid-state circuitry mounted on 7.7 by 14.1-inch printed wiring board (TN-type)
- Color coded faceplate labels identify the CP type and function (White = Control, Purple = Port or System Resource)
- Individual circuit functions all contained on one CP
- Metal tab for grounding
- Locking tab-type handle provides easy insertion or removal of a CP
- Port CPs can be inserted or removed with power “On” and the system processing calls. Only the calls utilizing circuits on a removed CP will be affected.

  Note: Power must be turned off when replacing the CPU/MEM or Service Circuit.

- Status LEDs

  — Port CPs:

  Red - “on’” several seconds during power up and test, “Off” with test pass. After test pass, “On” if fault in CP is subsequently detected.

  Green - “On” indicates resource available (port is translated).

  Yellow - “On” indicates a call in progress. “Off” when not in use.

  All LEDs “Off” - CP is not translated.

  — CPU/MEM CP: Green status LED only. “Off” for several seconds during power up and test, then lamp flashes to indicate an “OK” state. Steady “Off” or “On” indicates a problem.

— System Resource CPs:

Service Circuit CP - Similar to port CPs except yellow LED flashes to show system clock is active and is steadily “On” when a tone receiver is in use. “Off” indicates a problem.

Modem Pool and Tone Detector CPs - Same as Port CPs.

Terminal Equipment

Terminal equipment is connected to System 25 station (voice or data) ports. It is made up of the following groups:

- Voice Terminals
  - Single-Line
  - Multiline (MERLIN® Communications System sets and MET sets)
- Voice Terminal Adjuncts
- Attendant Consoles
- Asynchronous Data Units (for interface with data terminals).

This subsection provides information on all components in each group or contains references to the Section where information can be found.

Voice Terminals

System 25 supports a wide range of voice terminals, including industry standard touch-tone single-line sets and MERLIN System multiline sets.

In addition to providing basic telephone service (placing and answering calls), voice terminals can also be used to activate many system features. The voice terminals supported by System 25 are listed in Table 4-D and described in individual subsections.

Table 4-D. Summary of Voice Terminals

TERMINAL TYPE *	MODEL	DESCRIPTION	CIRCUIT PACK INTERFACE
Single-Line Tip Ring (Analog) †	420	Memory Set with Built-in Speakerphone	ZTN78 or TN742
	500MM	Rotary Desk Set	
	500SM	Rotary Desk Set Compatible with 4A Speakerphone	
	554BMPA	Rotary Wall Set	
	2500MMGB	Basic Touch-Tone Desk Set	
	2500MMGT	Basic Desk Set with Recall Button	
	2500DMGC	Basic Desk Set with Message Waiting	
	2500SM	Basic Desk Set Compatible with 4A Speakerphone	
	2514BMW	Basic Desk Set with Headset Jack	
	2526BMWG	Weatherproof Wall Set	
	2554BM	Basic Wall Set	
	7101A	Desk or Wall Set (MD)	
MERLIN Sys. Multiline 7300H Series (Hybrid)	7302H01D	5-Button	ZTN79
	7303H01D	10-Button	
	7305H01D	34-Button (MD)	
	7305H02D	34-Button Deluxe	
	7305H03B	BIS Set (Built-in Speakerphone)	
	7305H04C	BIS Set with Display	
	7309H01B	HFAI Set (Hands-Free-Answer on Intercom)	
	7313H01A	BIS-10 (10-Button)	
	7314H01A	BIS-22 (22-Button)	
	7316H01A	BIS-34 (34-Button)	
	7317H01A	BIS-34D (34-Button with Display)	
MET ‡ Multiline (Hybrid)	2991C05	10-Button (Desk)	TN735
	2991D05	10-Button (Wall)	
	2993C04	10-Button with BIS	
	7302M	12-Button (Desk)	

• System 25 supports several voice terminals that are no longer orderable. These include MET sets and the 34-button (basic) MERLIN System set.

• The system supports equivalent industry standard touch-tone single-line sets. Voice terminals connected via the ZTN78 Tip Ring Line CP must have a REN less than or equal to 1.20 A/B.

‡ The 2991C04 set [with Busy Lamp Field (BLF)] will not operate with System 25 unless specially modified. The BLF itself will always be inoperable.

Single-Line Voice Terminals

Single-line terminals can have only one incoming call ringing when the terminal is idle. The user at any busy single-line terminal can put an active call on hold and either originate a call or answer a waiting/camped-on call. Single-line terminals have access to most system features that do not require operation of programmable buttons.

All voice and control information between single-line terminals and the system digital switch is transmitted in analog form on tip and ring pairs of wire (one pair per terminal). Port circuits (ZTN78 Tip Ring Line CP or TN742 Analog Line CP) in the switch provide analog/digital conversion. Power for terminals is also sent from the switch on the tip and ring pairs. The pushbutton dials on single-line sets (except for the 500 Series) are touch-tone pads, which generate Dual Tone Multifrequency (DTMF) signals. The rotary 500 sets generate dial pulses.

The following subsections provide descriptions and illustrations of the single-line voice terminals supported by System 25.

500 Series

The Model 500 Series consists of conventional rotary dial telephones. They are recommended for use as a Power Failure Transfer (PFT) stations if the PFT trunk does not support touch-tone dialing. A 55A1 Ground Start button must be used with these sets if the PFT trunk is ground start. Rotary set users cannot do any procedures that require pressing the * or # buttons. The following 500 Series sets are supported by System 25; these sets are similar in appearance to the 2500 sets shown in Figure 4-4 except for their rotary dials.

- Model 500MM—Basic desk set (PEC 3100-0RD)
- Model 500SM—Desk set with 4A Speakerphone compatibility (PEC 3100-2RD)
- Model 554BMPA—Basic wall set (PEC 3100-0RW).

2500 Series

The following 2500 Series voice terminals have the following components and features:

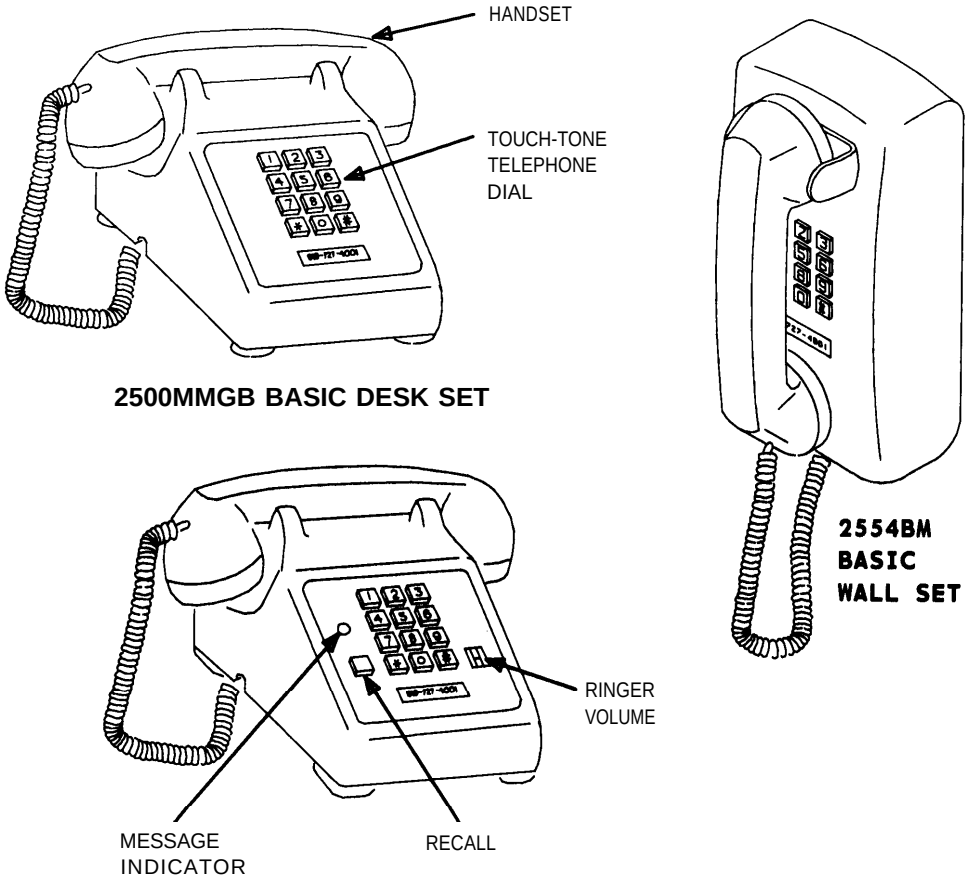
- Handset
- Touch-Tone Dial
- Ringer Volume Control.

Several 2500 Series voice terminals are shown in Figure 4-4. System 25 supports the following 2500 Series sets:

- Model 2500 DMGC—Desk Set with message waiting indicator and Recall button for timed switchhook flash (PEC 3178-SYS)

- Model 2500MMGB—Basic desk set (PEC 3100-1TD)
- Model 2500MMGT—Basic desk set with Recall button (PEC 3100-TRC)
- Model 2500SM—Basic desk set that can be used with a 4A Speakerphone (PEC 3100-2TD)
- Model 2514BMW—Basic desk set equipped with built-in headset jack
- Model 2554BM—Basic wall-mounted set (PEC 3100-TWR).

Adjuncts: Refer to Table 4-E.



2500MMGB BASIC DESK SET

2500DMGC DESK SET WITH EXTRA FEATURES

Figure 4-4. 2500 Series Analog Voice Terminals

Table 4-E. 2500 Series Voice Terminal Adjuncts

ADJUNCT	2500DMGC	2500MMGB	2500SM	2514BMW	2554BM
4A Speakerphone			x		
55A1 Ground Start Key	x	x	x	x	x
Answering Machine	x	x	x	x	x
Z34A Message Waiting Indicator (MD)		x	x	x	x

2526BMWG Voice Terminal

This analog terminal (Figure 4-5) consists of a standard touch-tone wall set equipped with a special faceplate and mounted in a weatherproof housing. The door of the housing can be fitted with an optional lock. This voice terminal is intended for outdoor use on buildings, fences, or poles. The 2526BMWG set can be connected for one line or two lines. It is approximately 13 inches high, 7 inches wide, and 6-1/2 inches deep.

The 2526BMWG set is not supplied fully assembled; each of the following parts must be ordered separately:

- 526A Housing
- 2526BMG Telephone Set Base (equipped with handset and cord)
- 253C Aluminum Faceplate
- D180352 Mechanical Door Lock (optional)
- D180849 Weathertight connecting arrangement for single-line service
- D180850 Weathertight connecting arrangement for two-line service
- D180805 Switch and arm bracket without weathertight arrangement.

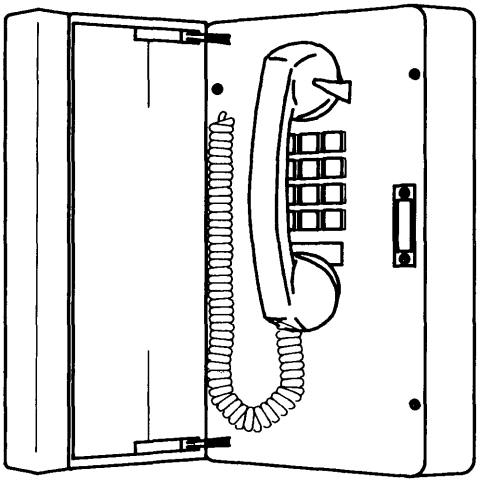


Figure 4-5. 2526BMWG Voice Terminal

7101A Voice *Terminal (MD) (PEC 3170-00M)*

The Model 7101 A single-line analog voice terminal (Figure 4-6) is about 2-3/4 inches wide, 3-1/2 inches high, and 8-1/2 inches deep. The set comes equipped with the following:

- Handset
- Touch-Tone Dial
- Message Indicator
- Tone Ringer with Volume Control
- Two Fixed Feature Buttons
  - — Recall - Used to place a call on hold and to obtain recall dial tone for Conference, Transfer, and other features accessible by feature access code.
  - — Disconnect - Used to disconnect one call and immediately obtain dial tone for another call.

Adjuncts: None

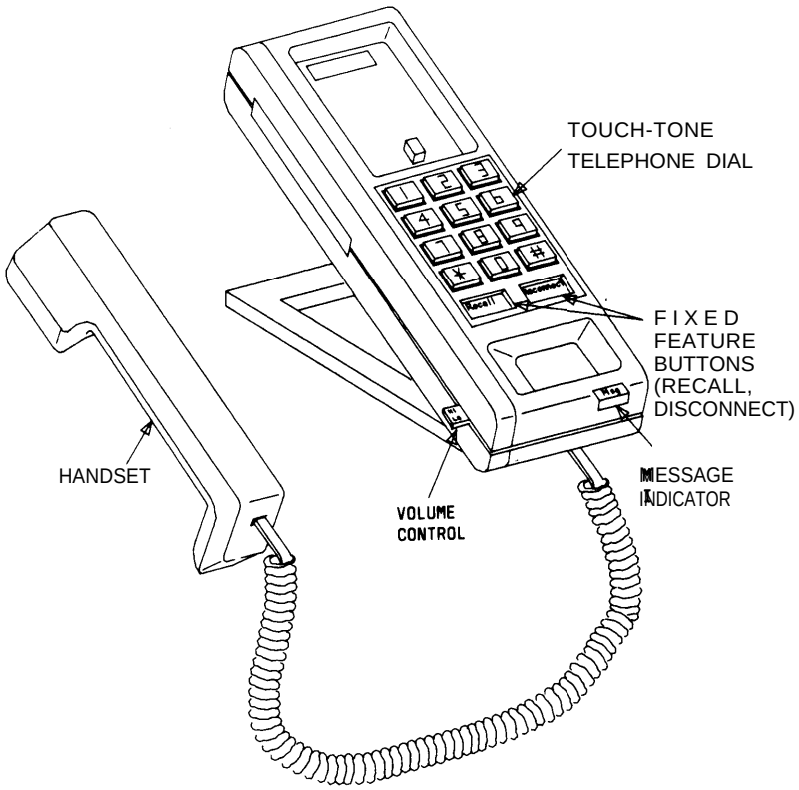


Figure 4-6. 710A Voice Terminal

420 Speakerphone Voice Terminal (Not Orderable)

The 420 Speakerphone voice terminal (Figure 4-7) is a single-line analog set that can be desk or wall mounted. The 420 Speakerphone set can no longer be ordered. It has the following components and features:

- Handset
- Touch-Tone Dial
- Built-In Speakerphone
- Twelve Memory Buttons (where emergency numbers and frequently called numbers can be stored for quick calling)
- Six Fixed Feature Buttons
  - — Program - For entering the memory button programming mode
  - — Redial - For recalling the last number dialed
  - — Flash - For generating a timed switchhook flash
  - — Mute - For turning off the speakerphone microphone temporarily for privacy
  - — Hold (with status LED) - For putting calls on hold
  - — Speaker (with status LED) - For making speakerphone calls and for turning on the speaker during handset calls
- Tone Ringer
- Three Volume Controls (tone ringer, speaker, handset receiver).

Adjuncts: None

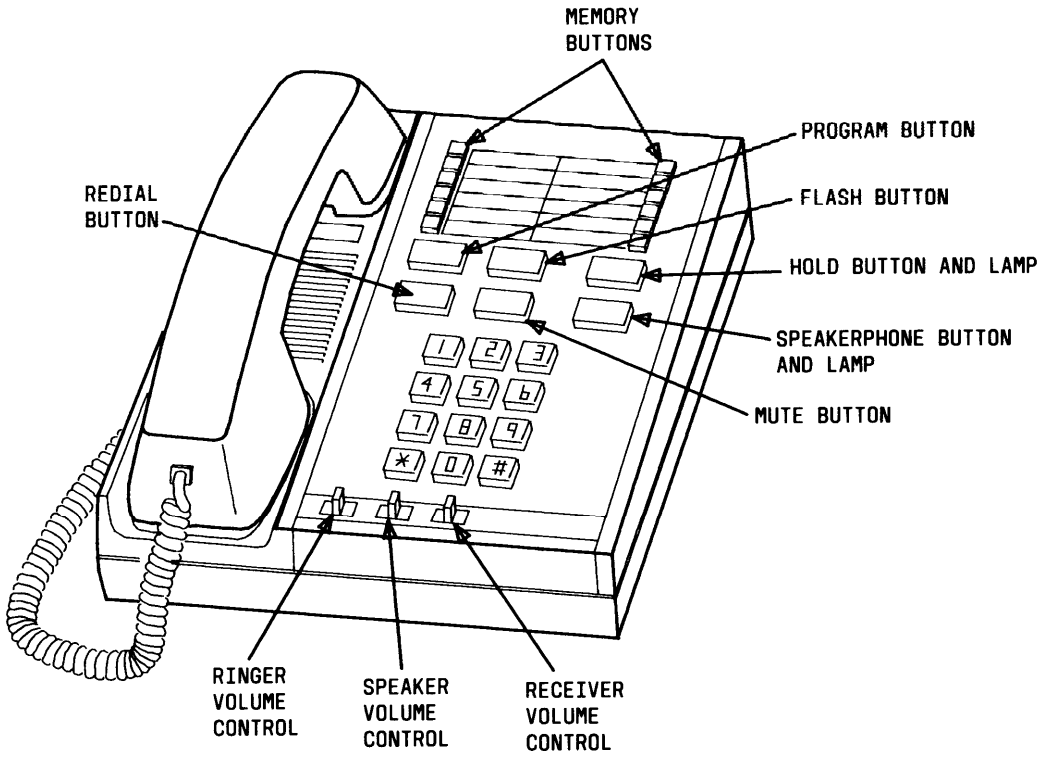


Figure 4-7. 420 Speakerphone Voice Terminal

Single-Line Voice Terminal Connection Information

Single-line voice terminal connection information is provided under the "Connectivity" heading later in this Section. Maximum cabling distances from the system cabinets to single-line voice terminals is provided in Section 5, "Technical Specifications."

Single-Line Voice Terminal Feature Operations

Refer to *Single-Line Terminal User Guide* (555-530-702) for information about single-line voice terminal feature operation.

Multiline Voice Terminals

The recommended multiline terminals for System 25 are the 7300H Series hybrid sets that are also used with the MERLIN Communications System. Multibutton Electronic Telephone (MET) sets already available to the customer can be reused in a System 25 installation but are not orderable.

Multiline voice terminals have programmable buttons that can be assigned for handling calls and for controlling features. Many of these buttons are supported by red l-Use and green status indicators (LEDs) that provide users with information about calls and features. Fixed (non-programmable) feature buttons allow users to control standard features such as Hold, Conference, and Transfer.

Multiline terminals can have several calls active at the same time depending on the number of buttons programmed for placing and receiving calls. For example, the user can be talking on one call, have one or more calls on hold, and still receive ringing on incoming calls.

Transmission to and from the terminals is hybrid: an analog pair for voice and two digital pairs for control signals. Port circuits (ZTN79 ATL Line CP for MERLIN System sets and TN735 MET Line CP for MET sets) provide interface between the terminals and the digital switch.

The pushbutton dials on MERLIN System terminals are not touch-tone; they send digital signals to the system switch and are referred to in terminal descriptions as a touch dial pads. MET sets have touch-tone (DTMF) dials.

The following subsections provide descriptions and illustrations of the multiline voice terminals supported by System 25.

7302H01D Voice Terminal (5-Button) (PEC 3160-111)

The 7302H01D voice terminal (Figure 4-8) can be desk or wall mounted and is about 5-3/4 inches wide, 5-1/4 inches high, and 8-1/2 inches deep. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-in Speaker
- Speaker/Ringer Volume Control
- Six Fixed Feature Buttons

Conference	Hold
Drop	Speaker
Transfer	Recall

- Five Programmable Feature Buttons (each equipped with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), and Last Number Dialed (l).

Adjuncts: None

Note: This set does not have a Message button or Message LED.

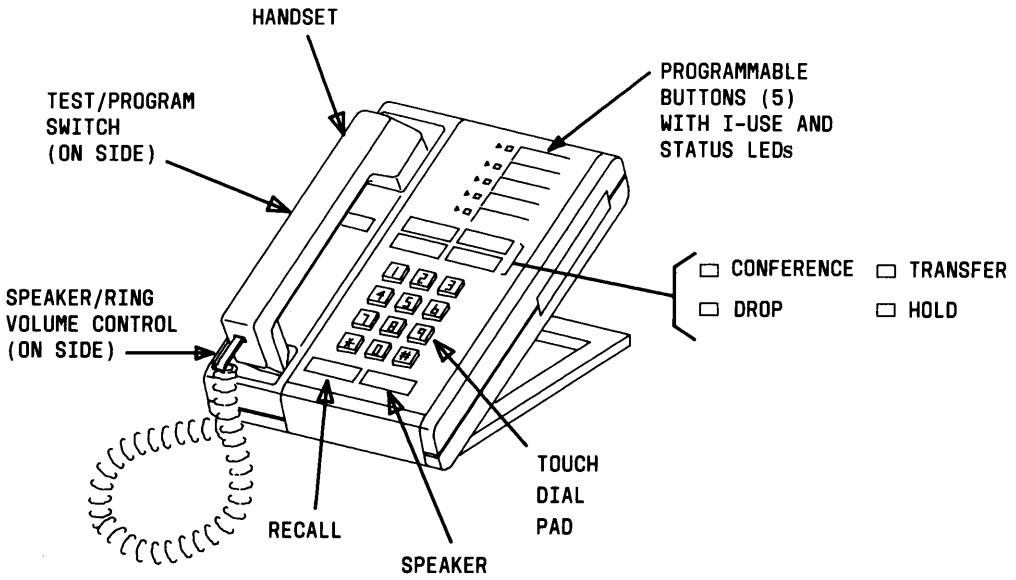


Figure 4-8. 7302H01D Voice Terminal (5-Button)

7303H01D Voice Terminal (10-Button) PEC 3161-172

The 7303H01D voice terminal (Figure 4-9) can be desk or wall mounted and is about 7 inches wide, 5-1/4 inches high, and 8-1/2 inches deep. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-in Speaker
- Speaker/Ringer Volume Control
- Seven Fixed Feature Buttons (* = with status LED)

Conference	Speaker*
Drop	Message*
Transfer	Recall
Hold	

- Ten Programmable Feature Buttons (each with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), Flex DSS (3), Send All Calls (l), Account Code Entry (1), and Last Number Dialed (l).

Adjuncts:

- 502A/B Headset Adapters

  Note: The 502B unit provides HFAI service on the headset.

- S102A Speakerphone (PEC 3163-HFU).

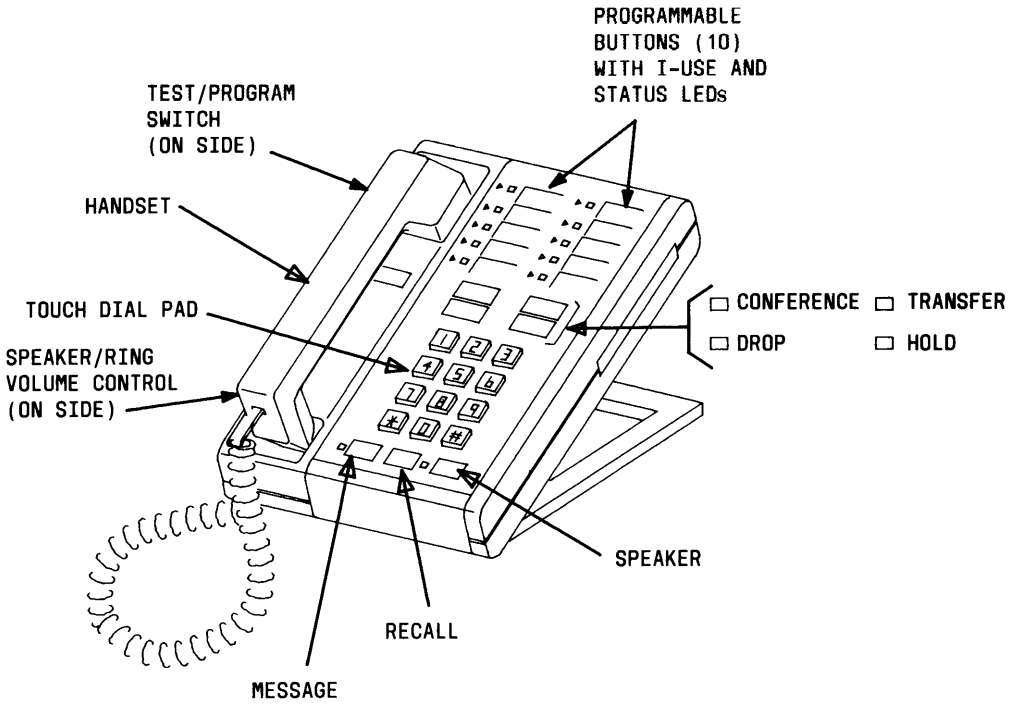


Figure 4-9. 7303H01D Voice Terminal (10-Button)

7305H01D Voice Terminal (34-Button) (MD) (PEC 3162-412)

The 7305 H01D voice terminal (Figure 4-10) can be desk or wall mounted and is about 10-1/4 inches wide, 5-1/2 inches high, and 8-1/2 inches deep. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-in Speaker
- Speaker/Ringer Volume Control
- Seven Fixed Feature Buttons (* = with status LED)

Conference	Speaker*
Drop	Message*
Transfer	Recall
Hold	

- 34 Programmable Feature Buttons (only ten with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), Flex DSS (27), Send All Calls (1), Account Code Entry (1), and Last Number Dialed (1).

  Note: Programmable buttons without LEDs should be used only for features that do not require l-Use and/or status indications.

Adjuncts:

- 502A/B Headset Adapters

  Note: The 502B unit provides HFAI service on the headset.

- S102A Speakerphone (PEC3163-HFU).

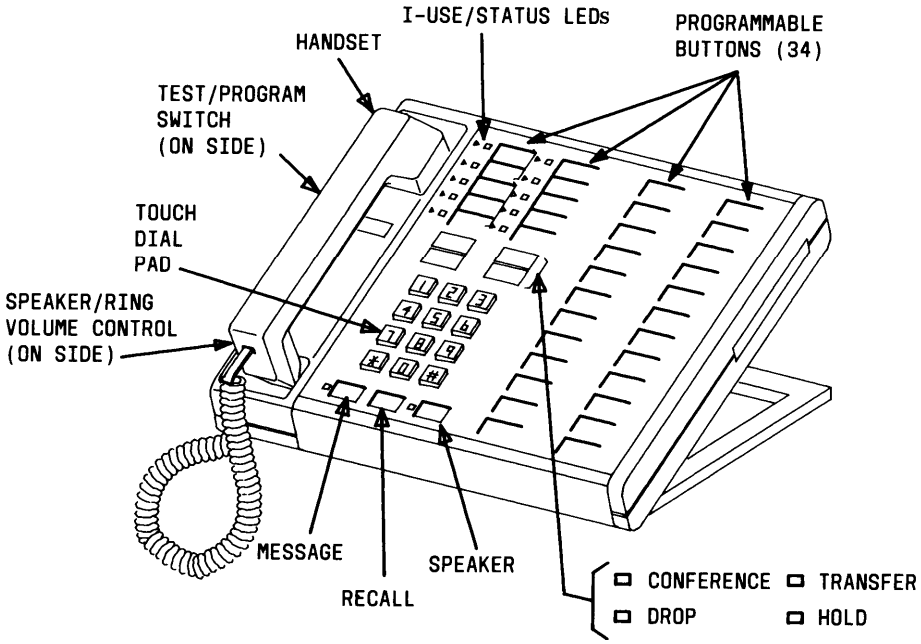


Figure 4-10. 7305H01D Voice Terminal (34-Button) (MD)

7305H02D Voice Terminal (34-Button Deluxe) (PEC 3162-417)

The 7305H02D voice terminal (Figure 4-11) is available for general use and as a Direct Trunk Attendant Console. The voice terminal is about 10-1/4 inches wide, 5-1/2 inches high, and 8-1/2 inches deep. It comes equipped with the following:

- Handset
- Touch Dial Pad (not DTIMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-in Speaker
- Speaker/Ringer Volume Control
- Seven Fixed Feature Buttons (* = with status LED)

Conference	Speaker*
Drop	Message*
Transfer	Recall
Hold	

- 34 Feature Buttons (each equipped with l-Use and Status LEDs)

  For General Use:

  — All programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (27), Send All Calls (1), Account Code Entry (1), and Last Number Dialed (l).

  For Use as a Direct Trunk Attendant Console:

  — Six predefine: Start, Cancel, Release, Return-On-Don't-Answer, Return-On-Busy, and Alarm.

  — Other 28 programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (1), Account Code Entry (1), Attendant Message Waiting (1), Night Service (1), trunk appearances (15 as Personal Lines**), Group Call Coverage (1), Direct Facility Access† (3), and Last Number Dialed (l).

  ** On the first Attendant Console, the first 15 trunks in the system are assigned button appearances on the console. If there are fewer than 15 trunks, the remaining buttons are not assigned. On the second Console, these trunks do not receive default assignments.

† On the first Attendant Console, the first of the Direct Facility (Pooled) Access buttons defaults to loop-start trunks, the second to ground-start trunks, and the third to tie trunks. For any trunk type not assigned in the system, the associated button does not receive a default assignment. On the second Console, these buttons do not receive default assignments.

Adjuncts:

- 502A/B Headset Adapters

  Note: The 5026 unit provides HFAI service on the headset.

- S102A Speakerphone (PEC 3163-HFU).

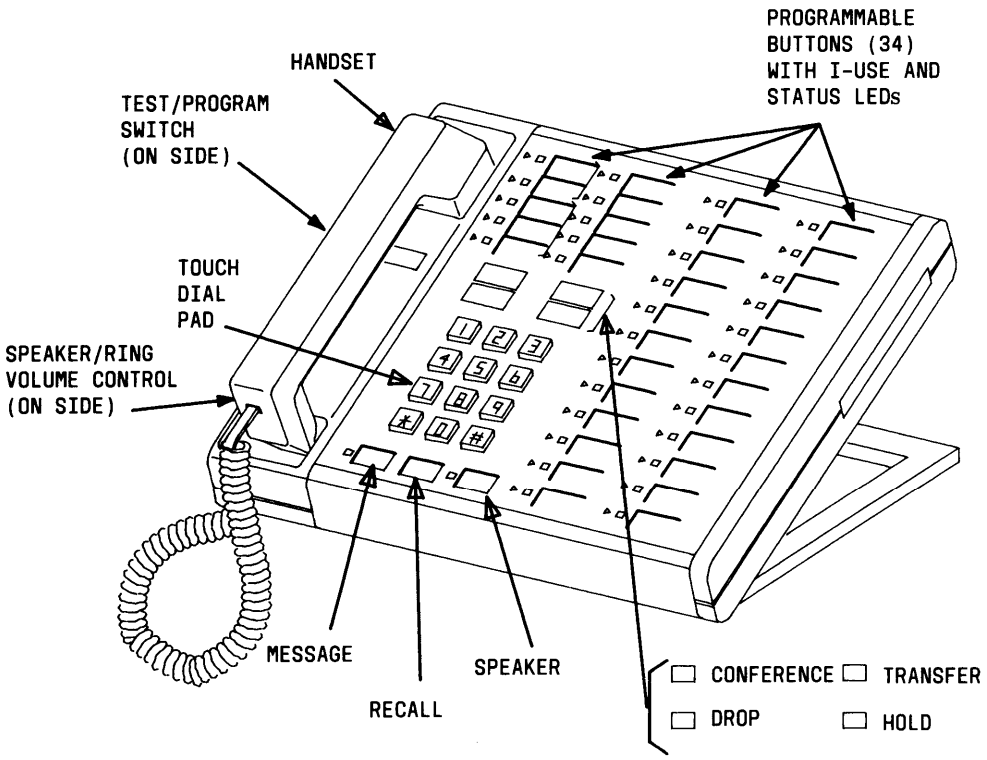


Figure 4-11. 7305H02D Voice Terminal (34-Button Deluxe)

7305H03B Voice Terminal (BIS) (PEC 3162-BIS)

The 7305H03B voice terminal (Figure 4-12) is available for general use and as a Direct Trunk Attendant Console. It can be desk or wall mounted and is about 9-1/2 inches wide, 5-1/4 inches high, and 9-1/4 inches deep. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-In Speakerphone (BIS) with Volume Control
- Built-In HFAI Microphone
- Ringer Volume Control
- Nine Fixed Feature Buttons (* = with status LED)

Conference	Recall
Drop	Message*
Transfer	Microphone*
Hold	HFAI*
Speakerphone*	

- 34 Programmable Feature Buttons (Only 12 have associated LEDs; programmable buttons without LEDs should be used only for features that do not require l-Use and Status indications.)

  For General Use:

  — All programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (27), Send All Calls (1), Account Code Entry (1), and Last Number Dialed (l).

  For Use as a Direct Trunk Attendant Console:

  — Six predefine: Start, Cancel, Release, Return-On-Don't-Answer, Return-On-Busy, and Alarm.

  — Other 28 programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (1), Account Code Entry (l), Attendant Message Waiting (1), Night Service (1), trunk appearances (15 as Personal Lines"), Group Call Coverage (1), Direct Facility Access† (3). and Last

Number Dialed (1).

** On the first Attendant Console, the first 15 trunks in the system are assigned button appearances on the console. If there are fewer than 15 trunks, the remaining buttons are not assigned. On the second Console, these trunks do not receive default assignments.

† On the first Attendant Console, the first of the Direct Facility (Pooled) Access buttons defaults to loop-start trunks, the second to ground-start trunks, and the third to tie trunks. For any trunk type not assigned in the system, the associated button does not receive a default assignment. On the second Console, these buttons do not receive default assignments.

Adjunct:

- 502A/B Headset Adapters

Note: The 502B unit provides HFAI service on the headset.

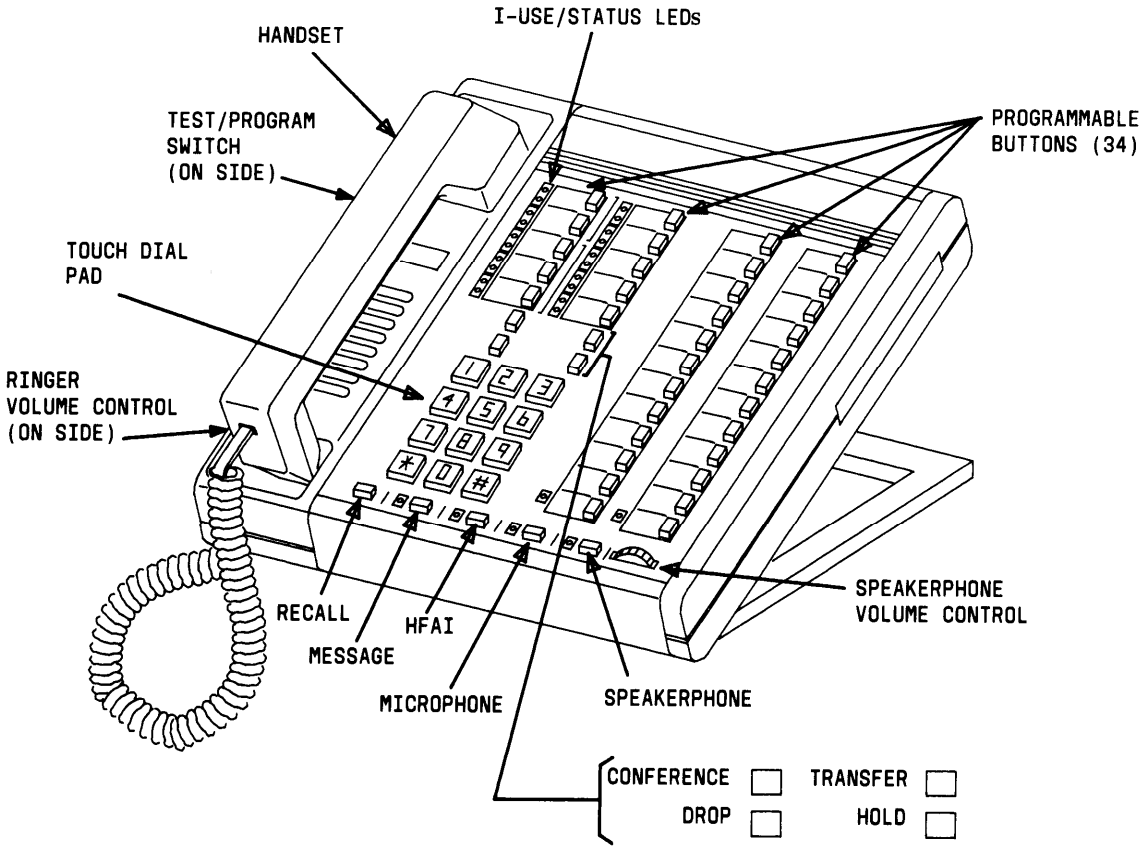


Figure 4-12. 7305H03B Voice Terminal (BIS)

7305H04C Voice Terminal (BIS With Display) (PEC 3162-DIS)

The 7305H04C voice terminal (Figure 4-13) is available for general use and as a Switched Looped Attendant Console (SLAC). Display capability can be administered for SLACs or general use positions. In general use, this terminal can be desk or wall mounted; as a console, it is normally desk mounted. The terminal is about 9-1/4 inches wide, 9-1/4 inches deep, and (not including the handset) 1-1/2 inches thick; when desk mounted, it is about 5-1/4 inches high in the back. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-In Display Module
  - — Screen for call information and time displays
  - — Clock/Timer controls
  - — Contrast control for screen display
- Built-In Speakerphone with Volume Control
- Built-In HFAI Microphone
- Ringer Volume Control
- Nine Fixed Feature Buttons (* = with status LED)

Conference	Hold	Message*
Drop	Speakerphone*	Microphone*
Transfer	Recall	HFAI*

- 34 Programmable Feature Buttons (only 12 have associated LEDs);

  For General Use:

  - — All programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (27), Send All Calls (1), Account Code Entry (1), and Last Number Dialed (l).

  For Use as a Switched Loop Attendant Console:

  - — Five predefine as loop buttons.

— Twelve others predefined as Inspect, Attendant Message Waiting, Alarm, Local, Scroll, Forced Release, Start, Source, Release, Destination, Cancel, and Join.

— Other seventeen programmable; default assignments are Flex DSS (15), Position Busy (1), and Last Number Dialed (1).

Note: Programmable buttons without LEDs should be used only for features that do not require I-Use and Status indications.

Adjunct:

- 502A/B Headset Adapters

Note: The 502B unit provides HFAI service on the headset.

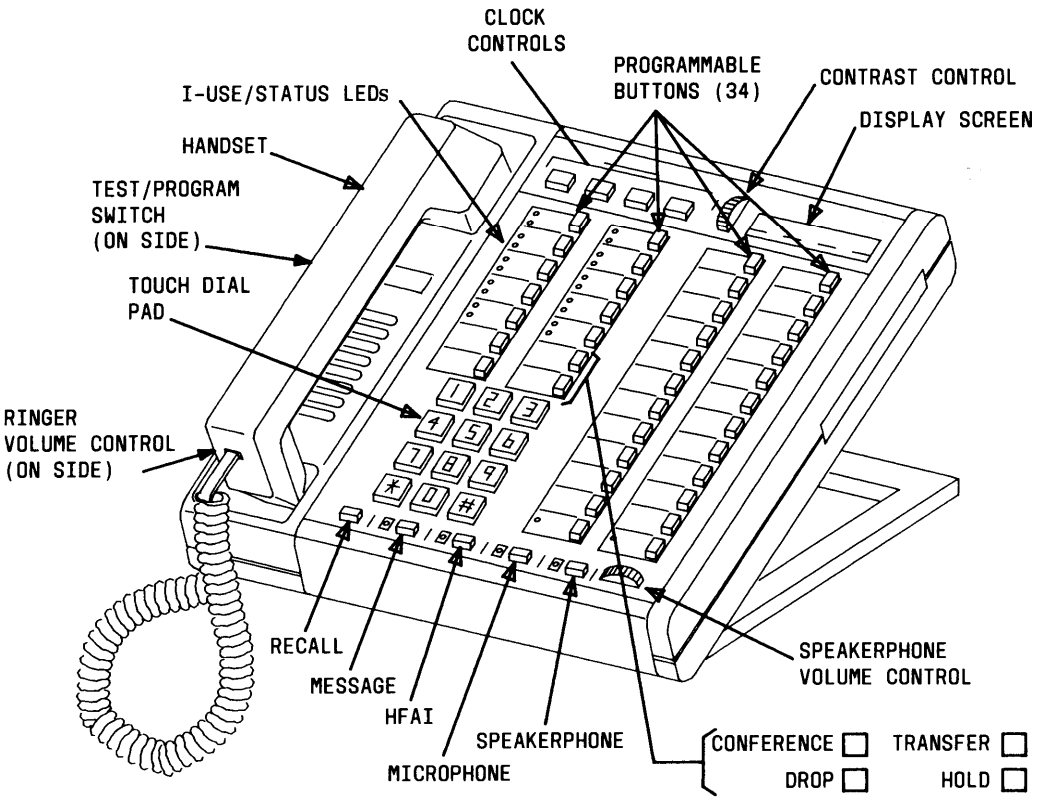


Figure 4-13. 7305H04C Voice Terminal (BIS With Display)

7309H01B Voice Terminal (HFAI) (PEC 3161-161)

The 7309H01B voice terminal (Figure 4-14) can be desk or wall mounted. It is about 6-1/4 inches wide, 8-3/4 inches deep, and (not including the handset) 1-1/2 inches thick; when desk mounted, it is about 5-1/4 inches high in the back. The set comes equipped with the following:

- Handset
- Touch dial pad (not DTMF)
- l-Use and Status LEDs
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-in HFAI Microphone
- Built-In Speaker
- Speaker/Ringer Volume Control
- Eight Fixed Feature Buttons (* = with status LED)

Conference	Speaker
Drop	Recall
Transfer	Message*
Hold	HFAI Microphone*

- Ten Programmable Feature Buttons (each equipped with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), Flex DSS (3), Send All Calls (l), Account Code Entry (l), and Last Number Dialed (l).

Adjuncts: None

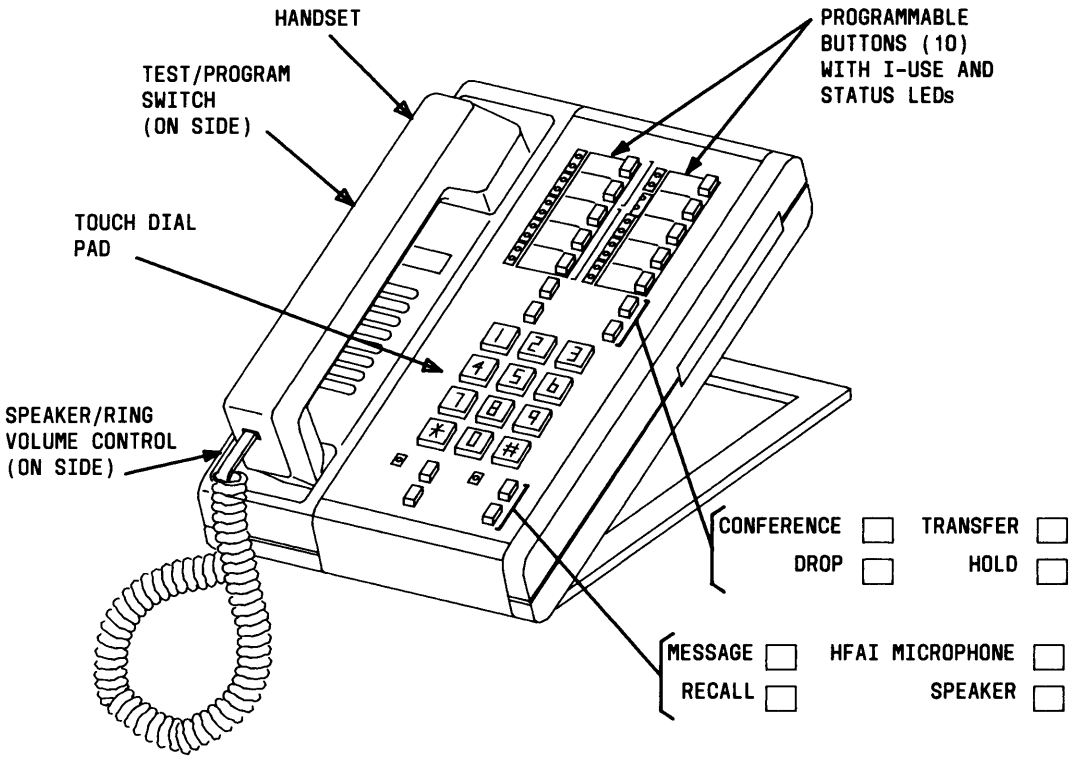


Figure 4-14. 7309H01B Voice Terminal (HFAI)

7313H01A Voice Terminal (BIS-10) (PEC 3165-10B)

This 7313H01A 10-button terminal (Figure 4-15) can be desk or wall mounted. It is about 6-1/4 inches wide, 8-3/4 inches deep, and (not including the handset) 1-1/2 inches thick; when desk mounted, it is about 5-1/4 inches high in the back. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- l-Use and Status LEDS
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-In Speakerphone With Volume Control
- Ringer Volume Control
- Eight Fixed Feature Buttons (* = with status LED)

Conference	Speakerphone*
Drop	Recall
Transfer	Message*
Hold	HFA1/Mic*

- 10 Programmable Feature Buttons (each equipped with l-Use and status LEDS); default assignments are System Access (2), Repertory Dialing (2), Flex DSS (3), Send All Calls (l), Account Code Entry (l), and Last Number Dialed (l).

Adjunct:

- 502A/B Headset Adapters

Note: The 502B unit provides HFAI service on the headset.

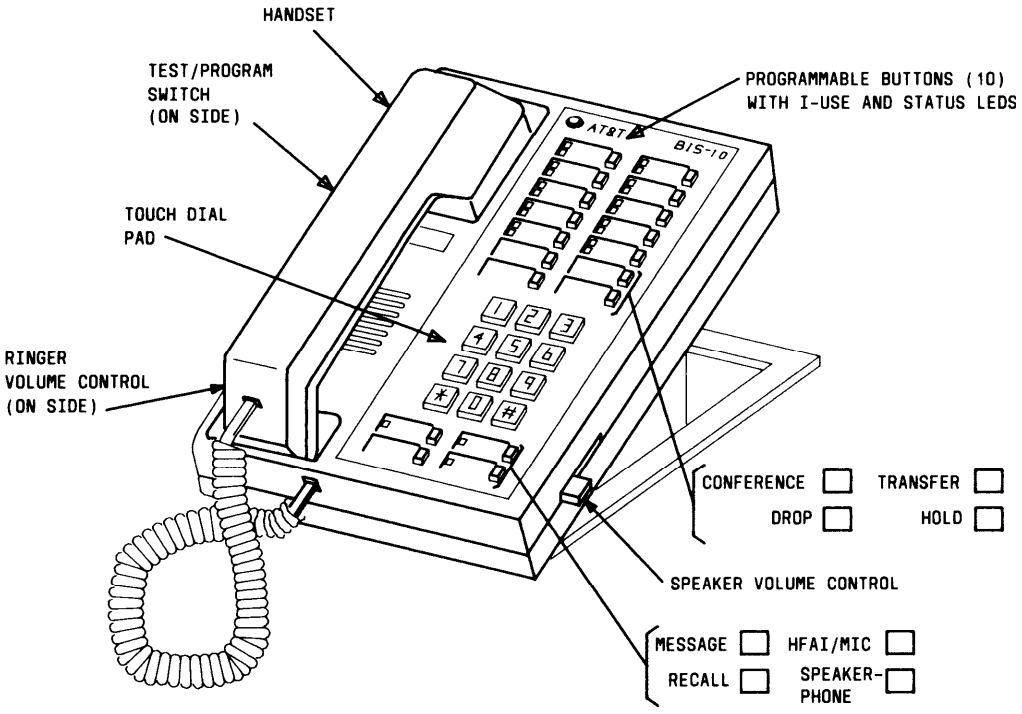


Figure 4-15. 7313H01A Voice Terminal (BIS-10)

7314H01A Voice Terminal (BIS-22) (PEC 3166-228)

The 7314H01A 22-button voice terminal (Figure 4-16) can be desk or wall mounted. It is about 8-1/4 inches wide, 9-1/4 inches deep, and (not including the handset) 1-1/2 inches thick; when desk mounted, it is about 5-1/4 inches high in the back. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- i-Use and Status LEDS
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-In Speakerphone With Volume Control
- Ringer Volume Control
- Nine Fixed Feature Buttons (* = with status LED)

Conference	Speakerphone*
Drop	Message*
Transfer	HFAI*
Hold	Mic*
Recall	

- 22 Programmable Feature Buttons (each equipped with l-Use and status LEDS); default assignments are System Access (2), Repertory Dialing (2), Flex DSS (15), Send All Calls (l), Account Code Entry (l), and Last Number Dialed (1).

Adjunct:

- 502A/B Headset Adapters

Note: The 502B unit provides HFAI service on the headset.

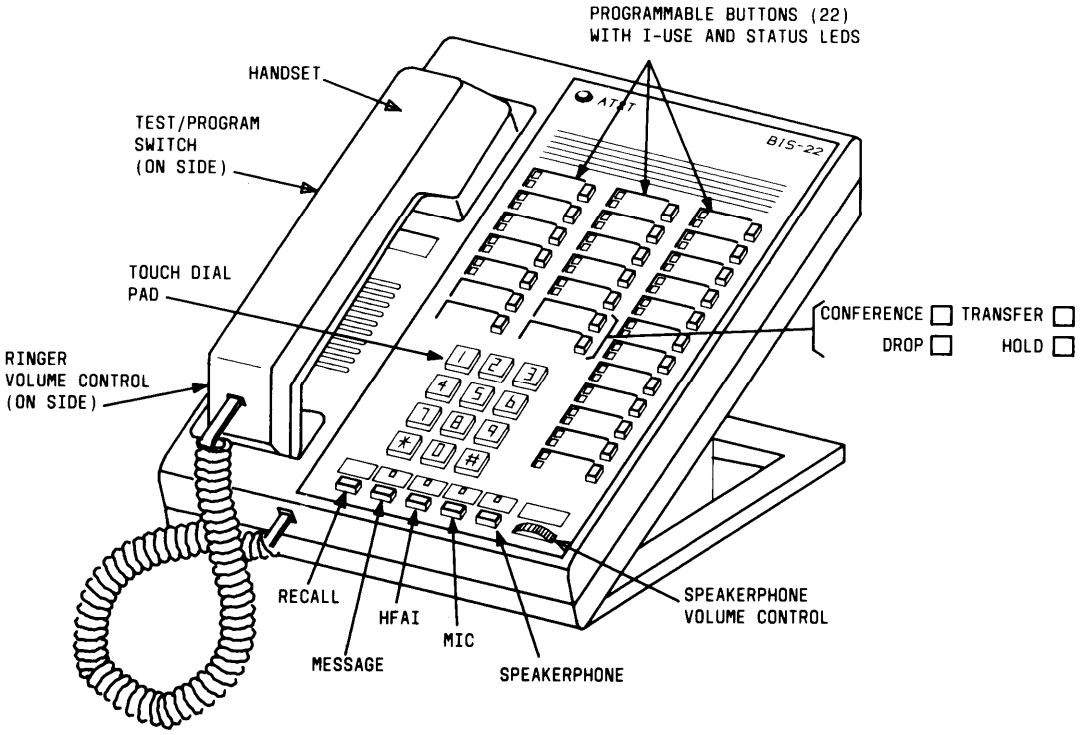


Figure 4-16. 7314H01A Voice Terminal (BIS-22)

7316H01A Voice Terminal (BIS-34) (PEC 3167-34B)

The 7316H01A 34-button voice terminal (Figure 4-17) is available for general use or as a Direct Trunk Attendant Console. It can be desk or wall mounted and is about 9-1/4 inches wide, 9-1/4 inches deep, and (not including the handset) 1-1/2 inches thick; when desk mounted, it is about 5-1/4 inches high in the back. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDS
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-In Speakerphone With Volume Control
- Ringer Volume Control
- Nine Fixed Feature Buttons (* = with status LED)

Conference	Speakerphone*
Drop	Message*
Transfer	HFAI*
Hold	Microphone*
Recall	

- 34 Programmable Feature Buttons (each equipped with I-Use and status LEDS);

  For General Use:

  — All programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (27), Send All Calls (1), Account Code Entry (1), and Last Number Dialed (1).

  For Use as a Direct Trunk Attendant Console:

  — Six predefined: Start, Cancel, Release, Return-On-Don't-Answer, Return-On-Busy, and Alarm.

  — Other 28 programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (1), Account Code Entry (1), Attendant Message Waiting (1), Night Service (1), trunk appearances (15 as Personal Lines**), Group Call Coverage (1), Direct Facility Access† (3), and Last Number Dialed (1).

** On the first Attendant Console, the first 15 trunks in the system are assigned button appearances on the console. If there are fewer than 15 trunks, the remaining buttons are not assigned. On the second Console, these trunks do not receive default assignments.

† On the first Attendant Console, the first of the Direct Facility (Pooled) Access buttons defaults to loop-start trunks, the second to ground-start trunks, and the third to tie trunks. For any trunk type not assigned in the system, the associated button does not receive a default assignment. On the second Console, these buttons do not receive default assignments.

Adjunct:

502A/B Headset Adapters

Note: The 502B unit provides HFAI service on the headset.

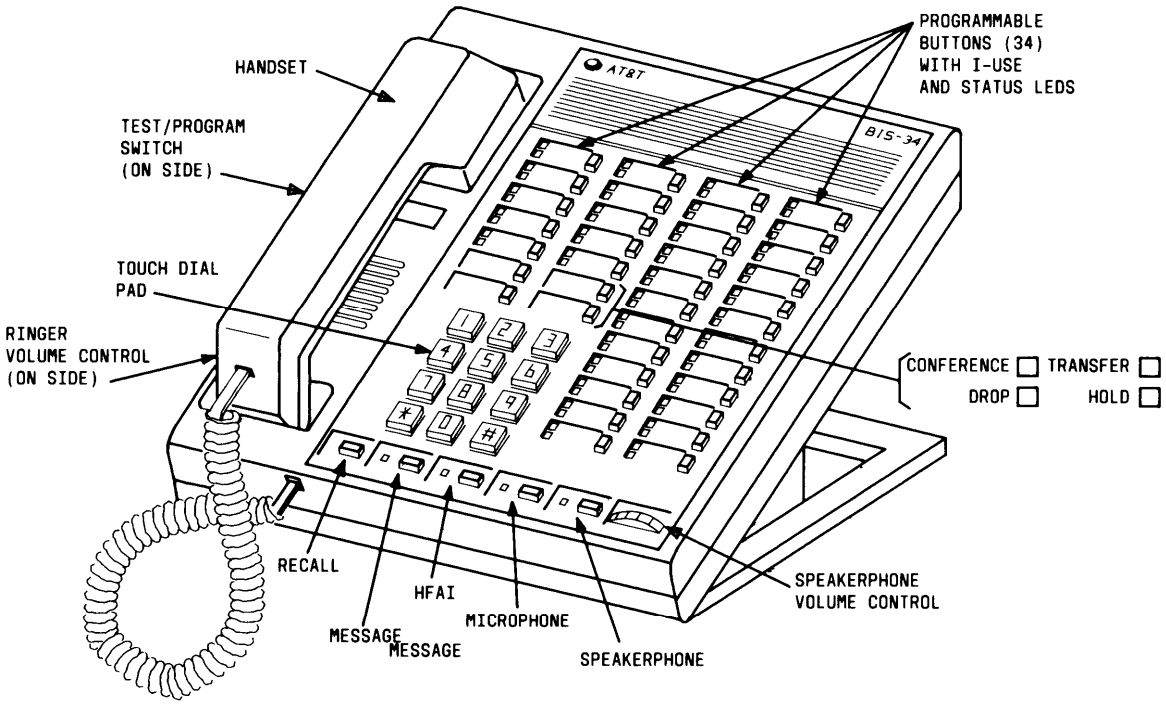


Figure 4-17. 7316H01A Voice Terminal (BIS-34)

7317H01A Voice Terminal (BIS-34D) (PEC 3167-DSB)

The 7317H01A 34-button voice terminal with display (Figure 4-18) is available for general use and as a SLAC. In general use, it can be desk or wall mounted; as a console, it is normally desk mounted. Display can be administered for SLACs and general use positions. The set is about 9-1/4 inches wide, 9-1/4 inches deep, and (not including the handset) 1-1/2 inches thick; when desk mounted, it is about 5-1/4 inches high in the back. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDS
- Test/Program Switch (two positions: T for testing operation of LEDs and ringer; P for programming feature buttons)
- Built-In Display Module
  - — Screen for call information and time displays
  - — Clock/Timer controls
  - — Contrast control for screen display
- Built-In Speakerphone With Volume Control
- Ringer/Button Click Volume Control
- Built-In HFAI Microphone
- Nine Fixed Feature Buttons (* = with status LED)

Conference	Hold	Message*
Drop	Recall	HFAI*
Transfer	Speakerphone*	Microphone*

- 34 Feature Buttons (each equipped with I-Use and status LEDS)

  For General Use:

  - — All programmable; default assignments are System Access (2), Repertory Dialing (2), Flex DSS (27), Send All Calls (1), Account Code Entry (1), and Last Number Dialed (1).

  For Use as a Switched Loop Attendant Console:

  - — Five predefined as loop buttons.

— Twelve others predefined as Alarm, Source, Destination, Inspect, Local, Cancel, Start, Scroll, Forced Release, Attendant Message Waiting, Join, and Release.

— Other seventeen programmable; default assignments are Flex DSS (15), Position Busy (1), and Last Number Dialed (1).

Adjunct:

- 502A/B Headset Adapters

Note: The 502B unit provides HFAI service on the headset.

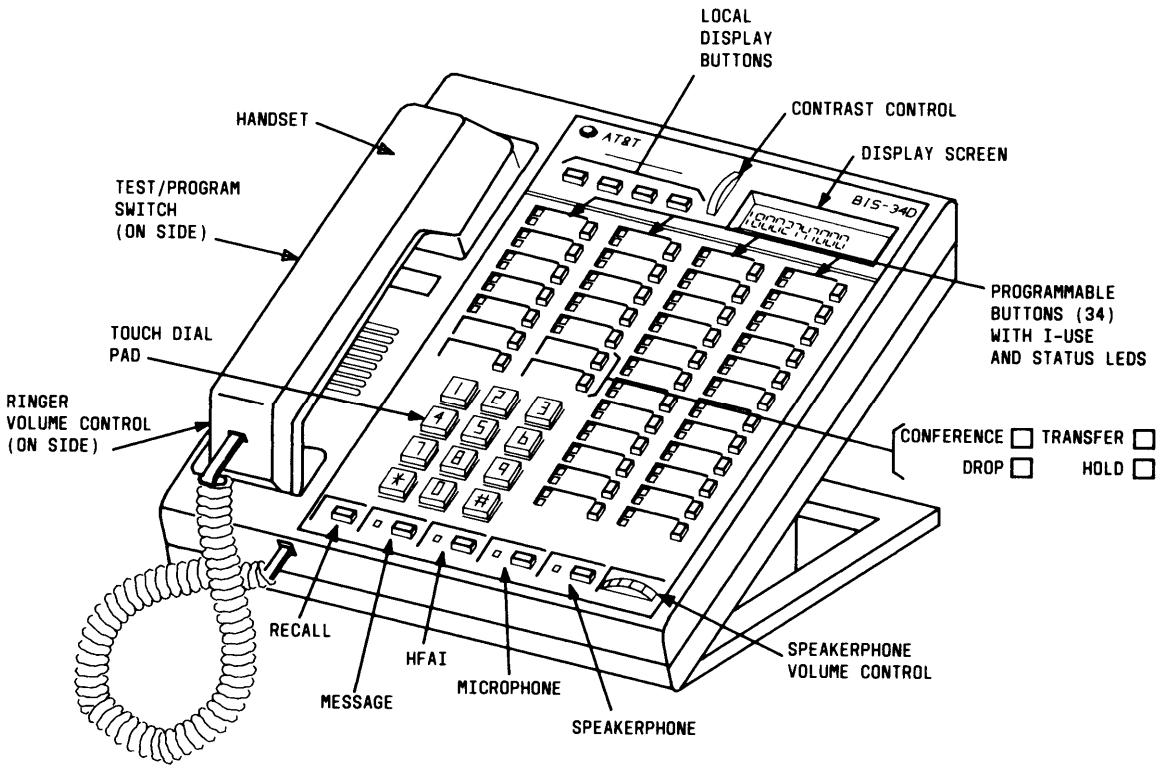


Figure 4-18. 7317H01A Voice Terminal (BIS-34D)

10-8utton MET Set (2991 C/D05)

The 10-Button MET set (Figure 4-19) may be desk or wall mounted. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Contigurator.* The set comes equipped with the following:

- Handset
- Touch-Tone Dial Pad (DTMF)
- l-Use and Status LEDs
- Tone Ringer Volume Control
- Six Fixed Feature Buttons

Recall	Transfer
Conference	Hold
Drop	Message

- Five Programmable Feature Buttons (each equipped with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), and Last Number Dialed (1).

Adjuncts:

- 4A Speakerphone
- MET Headset Adapter.

Note: The Busy Lamp Field (BLF) version of this set, unless modified, will not work on System 25.

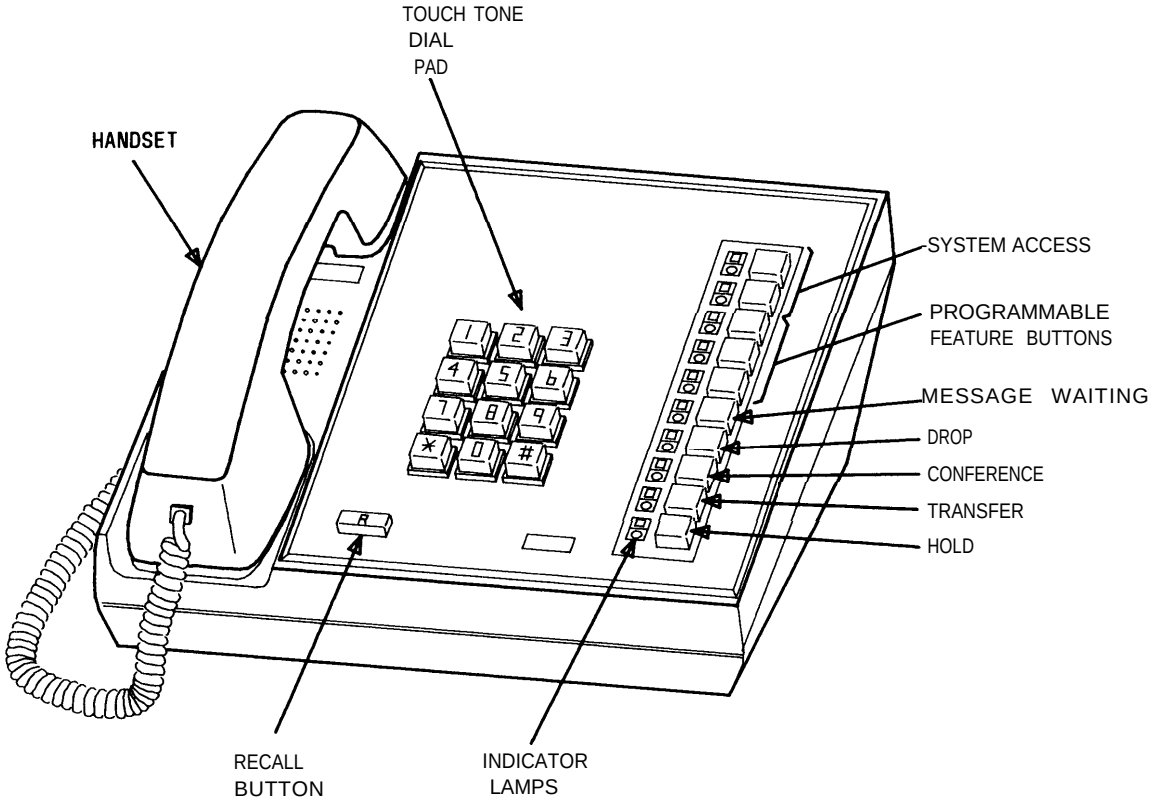


Figure 4-19. 10-Button MET Set (2991C05)

10-Button MET Set With Built-In Speakerphone (2993C04)

The 10-Button MET set with BIS (Figure 4-20) can be desk or wall mounted. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch dial pad (DTMF)
- l-Use and Status LEDs
- Built-In Speakerphone
- Tone Ringer Volume Control
- Speakerphone Volume Control
- On/Quiet and Off Speakerphone Control Buttons
- Speakerphone Indicator Lamp
- Six Fixed Feature Buttons

Recall	Transfer
Conference	Hold
Drop	Message

- Five Programmable Feature Buttons (each equipped with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), and Last Number Dialed (1).

Adjuncts: None

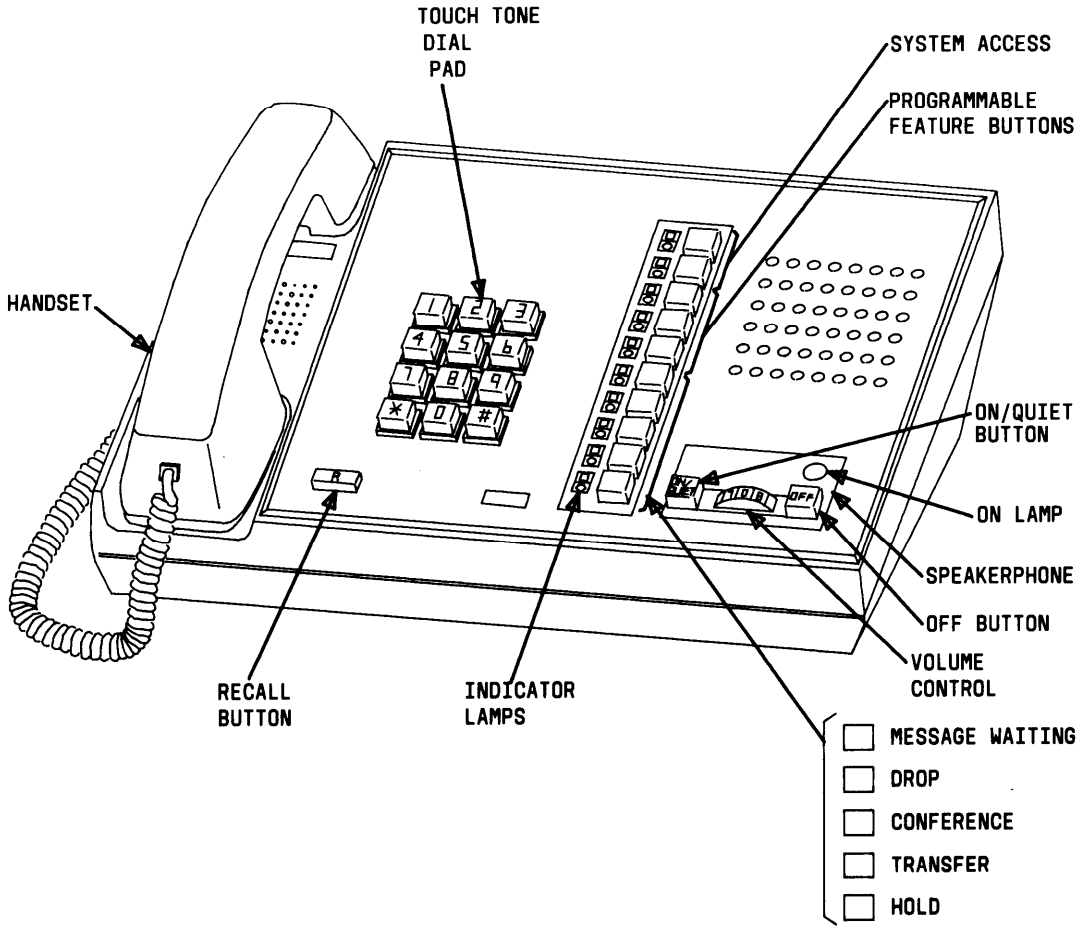


Figure 4-20. 10-Button MET With Built-In Speakerphone (2993C04)

12-Button MET Set (7203M)

The 12-Button MET set (Figure 4-21) is a freestanding voice terminal. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch-Tone Dial Pad (DTMF)
- l-Use and Status LEDs
- Tone Ringer Volume Control
- Message Waiting LED
- Seven Fixed Feature Buttons

Recall	Transfer
Conference	Hold
Drop	Message
Disconnect	

- Five Programmable Feature Buttons (each equipped with l-Use and Status LEDs); default assignments are System Access (2), Repertory Dialing (2), and Last Number Dialed (1).

Adjuncts:

- 500A Headset Adapter
- S101A Speakerphone.

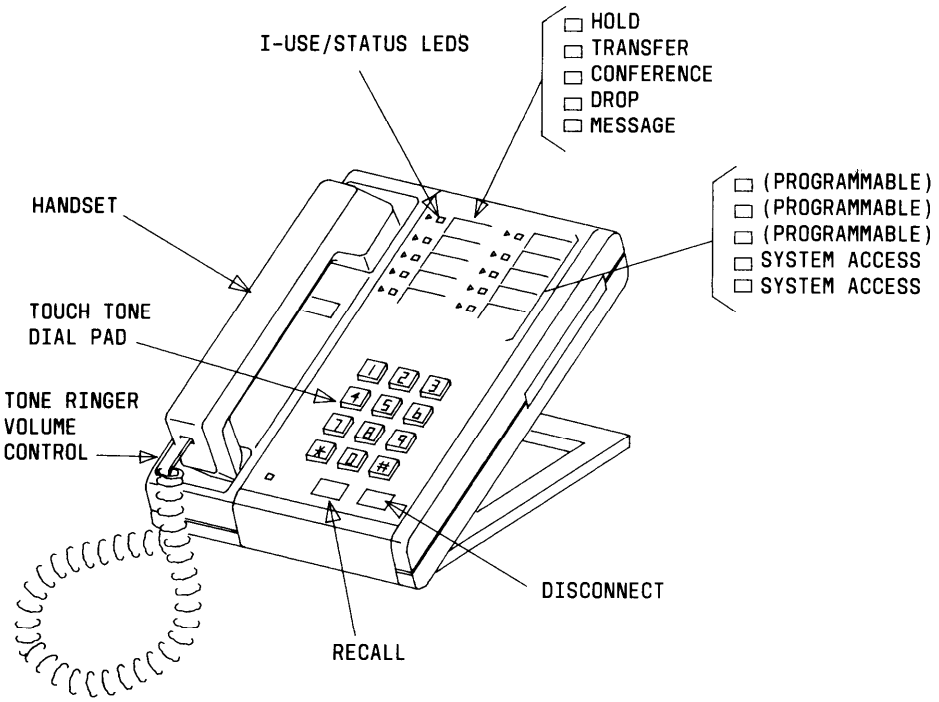


Figure 4-21. 12-Button MET Set (7203M)

Multiline Voice Terminal Connection information

Detailed connection information is provided under the "Connectivity" heading later in this Section. Maximum cabling distances from the system cabinets to multilane voice terminals is provided in Section 5, "Technical Specifications."

Multiline Voice Terminal Feature Operations

Refer to *Multiline Terminal User Guide* (555-530-703) for information about feature operation.

Voice Terminal Adjuncts

The following adjuncts and associated power supplies are supported on multiline sets:

- MET Headset Adapter (for 10-Button MET sets); refer to “Headset Adapter Adjunct” in Section 2.
- 500A/502A/502B Headset Adapter (for 12-Button MET sets and MERLIN System voice terminals); refer to “Headset Adapter Adjunct” in Section 2. The 5026 must be used if Hands-Free Answer on Intercom (HFAI) operation on the headset is desired (for example, typical Call Management System application).
- 4A Speakerphone System (for 2500SM single-line sets and 10-Button MET sets); refer to “Speakerphone Adjunct” in Section 2.
- S101A/S102A Speakerphone (for 12-Button MET sets and MERLIN System voice terminals, respectively); refer to “Speakerphone Adjunct” in Section 2.
- 55A1 Key (Ground Start Button)

  A 55A1 Ground Start Button is required for each Power Failure Transfer (PFT) station that is connected to a ground start trunk during power failures.

- Acoustic Coupler

  An Acoustic Coupler (349A Adapter) can be used with the 7300H series voice terminals and MET sets. The coupler provides acoustic coupling between the handset and acoustic modems.

- Answering/Announcement Machine

  A suitable answering/announcement machine can be used as an adjunct to single-line voice terminals. Note that when such a machine is bridged on to a ZTN78 Tip Ring Line CP port, the combined adjunct/terminal REN must not exceed 1.20 A/B.

- Z34A Message Waiting Indicator (MD)

  The Z34A Message Waiting Indicator provides a message waiting light at 2500 Series single-line voice terminals that do not have a built-in lamp. Existing sets with this adjunct can be reused.

Voice Terminal Adjunct Connection Information

Detailed adjunct connection information is provided in Section 2 with the detailed feature descriptions of the headset adapter and speakerphone adjuncts.

Descriptions of the Station Interconnect Panel (SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under the “Connectivity” heading later in this Section.

Voice Terminal Adjunct Power Supplies

Table 4-F provides a summary of the supplemental power supplies and their applications.

Table 4-F. Supplemental Voice Terminal Power Supplies

POWER SUPPLY	OUTPUT	FOR USE WITH
2012D Transformer	18V ac	ADUs (except Z3A5). MET sets that require local power. 500A Headset Adapter. S101A Speakerphone.
KS-22911, L1 Power supply	-48 V dc	Selector Console. Z3A5 ADU. 7300 H-Series sets that require local power. 502A/B Headset Adapter (see note). S102A Speakerphone (see note).
85B1 Power Unit	18V ac	4A Speakerphone System

Note: This power supply is required whenever an adjunct (Headset Adapter or Speakerphone) is connected to a 22- or 34-button voice terminal.

Attendant Consoles

System 25 can have one of the following attendant console configurations:

- One or two Direct Trunk Attendant Consoles (DTACs)—34-Button Voice Terminals (PEC 3162-417, 3162-BIS, or 3167-34B) administered for attendant service. Either or both positions can have a Direct Extension Selector Console associated with it.
- One or two Switched Loop Attendant Consoles (SLACs)—34-Button BIS/Display Voice Terminals (PEC 3162-DIS or 3167-DSB) administered for attendant service. Either or both positions can have a Direct Extension Selector Console associated with it.

Complete information on the Attendant Console features can be found in Section 2.

Asynchronous Data Units (ADUs)

Asynchronous Data Units and Multiple Asynchronous Data Units (MADUs) provide an interface between ports on the TN726 Data Line CP and RS-232 Data Terminal Equipment (DTE) or Data Communications Equipment (DCE). The DTE is equipment that provides a data source, termination, or both—a host computer, printer, or a data terminal are examples of DTE. The DCE is equipment that provides the functions required to establish, maintain, and terminate data communications—modems are the most common DCE.

The Z3A series of ADUs (Figure 4-22 and 4-36) are DCE that allow a direct connection between DTE and port circuits on the Data Line CP (TN726). To connect an ADU to DCE, a cross-over cable ("null modem") is required (PEC 2724-30 C).

The modular jack labeled "Wall" connects the ADU to the building wiring with a standard 4-pair modular cord. The 400B2 Adapter can be used to provide supplemental ac power for the ADU and is bridged at the wall jack if required (Z3A1, 2, and 4 units only).

The modular jack labeled "Telephone" allows a voice terminal to be attached to the ADU. Separate wire pairs from the telephone to the system cabinets are provided in a single 4-pair cable run back to the SIP. The pairs separate at the SIP for connection to voice and data ports.

The Z3A series of ADUs measure about 4.5 inches in length, 2 inches wide, and 1 inch high. The ADUs available are shown in Table 4-G.

The Z3A ADUs should be installed only on "inside" facilities; they are not designed to be used with CO cables or with exposed outside wiring (such as aerial cables).

The Z3A series of ADUs offer the following features:

- Provide an interface to the digital switch from RS-232 devices.
- Increases the distance RS-232 signals can travel over standard twisted-pair wiring. Refer to Section 5, "Technical Specification" for distance limitations.
- Data and control signals can be transmitted 2,000 feet in asynchronous full-duplex mode at speeds up to 19,200 bps. The transmission speed automatically matches that of the attached RS-232 device.
- The dc isolation via opto-couplers ensures high noise immunity, resulting in very low error rates.
- A variety of Z3As with different connectors allows easy connection to RS-232 terminals, printers, and host computers (see Table 4-G).
- Most Z3As can be powered from the RS-232 interface. The ADU requires 7 volts on pin 20 (DTR) to operate properly. If the RS-232 equipment cannot meet this requirement, a low-voltage power transformer and adapter(s) must be connected. Z3A5 ADUs *always* require supplemental power.

- An analog single-line voice terminal (2500 or 7100 series) or a 7300H series multiline voice terminal (Z3A5 ADU required) can be connected to the ADU, allowing the voice terminal and DTE to share a common wall jack and 4-pair cable run back to the SIP.

Note: Neither off-premises nor out-of-building service can be provided with ADUs.

For additional information on ADUs, see *Z3A Asynchronous Data Unit User Manual* (555-401-701).

ADU Connection Information

Detailed connection information is provided under the "Connectivity" heading later in this Section.

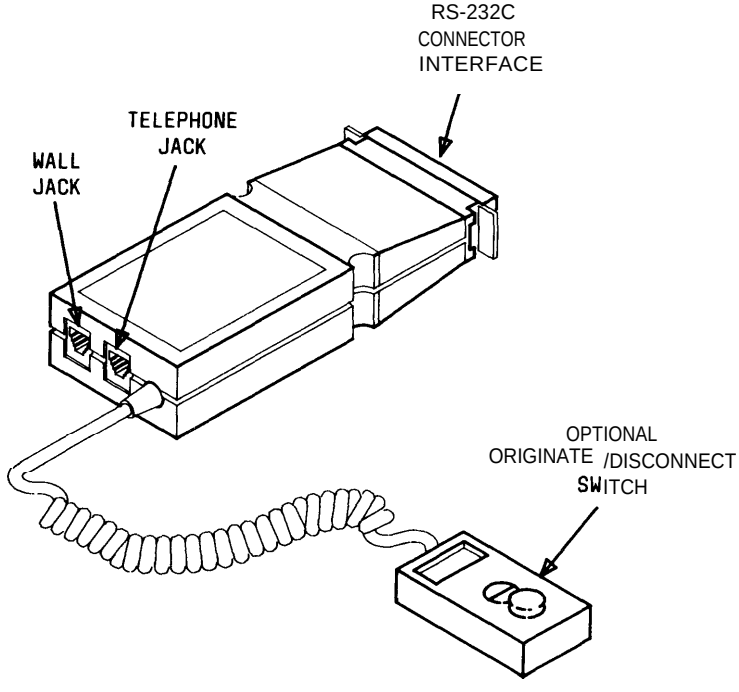


Figure 4-22. Asynchronous Data Unit (ADU)

Table 4-G. Asynchronous Data Units

UNIT	PEC	FEATURE
Z3A1	2169-001	3-foot plug-ended EIA connector and mod jack for single-line set.
Z3A2	2169-002	EIA plug and mod jack for single-line set.
Z3A4	2169-004	3-foot receptacle-ended EIA cord and mod jack for single-line set.
Z3A5	62506	3-foot plug-ended EIA connector and mod jack for hybrid set. (PEC includes KS-22911 L1 power supply and D8W-87 cord.)
MADU	2169-005	Self-powered. Used for host or protocol converter connections where voice terminals are not required. No sets can be connected directly to the MADU.

Peripheral Equipment

Peripheral Equipment includes the following devices that connect to the call processing portion of the CPU/MEM (ZTN129) CP:

- System Administration Terminal (SAT); refer to “System Administration” in Section 2.
- Digital Tape Unit (DTU); refer to “Digital Tape Unit” in Section 2.
- Station Message Detail Recording (SMDR) printer or Call Accounting System (CAS); refer to “Station Message Detail Recording” or “Call Accounting System” in Section 2.

Auxiliary Equipment

Auxiliary equipment supports System 25 features and services. The following equipment is supported:

- Dictation Equipment; refer to “Dictation System Access” in Section 2.
- External Alerting Equipment; refer to “External Alerts” in Section 2.
- Music Source (Music-On-Hold); refer to “Music-On-Hold” in Section 2.
- Paging Equipment; refer to "Paging System Access” in Section 2.
- Recorded Delay Announcement Equipment; refer to “Direct Group Calling Delay Announcement” and “Night Service Delay Announcements” in Section 2.
- Optional Power Equipment.

Note 1: Auxiliary equipment connected to the ZTN78 Tip Ring Line CP must meet the following requirements:

— AC impedance: 600 ohms

— DC current less than 30 ma at 24 V dc

— Ringer Equivalent Number (REN) less than 1.20

— Distance must not exceed 2000 feet

Note 2: Off-premises auxiliary equipment must be connected to the TN742 Analog Line CP. If the auxiliary equipment requires a contact closure, the TN763 Auxiliary Trunk CP must be used (on-premises service only).

Optional Power Equipment

In addition to the power supplies already mentioned, the following equipment can be used with System 25.

Uninterruptible Power Supply

The AT&T 1KVA Uninterruptible Power Supply (UPS) Model 010U111 PEC 2403-004 is recommended. At maximum load the UPS will bridge a 5-minute power outage. The UPS must be connected to the common System 25 power outlet. One UPS will support a 2-cabinet system.

AC Power Line Surge Suppressor

The Tll Model 428 Self-Restoring Power Line Surge Suppressor (PEC 8310-001, Comcode 402988950) protects against electrical surges, spikes, and transients that can cause damage to the System 25 power supply. A pilot light indicates that full protection is present. The unit plugs directly into a standard 120-volt 15-amp grounded outlet, providing a dual outlet for protected equipment.

346 Modular Bulk Power Supply

The 346 Modular Bulk Power Supply (346 MBPS) is a cost effective and flexible alternative to the KS-22911 power supply. The 346 MBPS can be used where the wall outlet mounted KS-22911 cannot be used (Canada) or where multiple KS units are required. The 346 MBPS consists of the 346A Power Unit (346A PU) and the 346A1 Power Panel; that is the sole method of mounting the power units. Up to three 346A PUs can be mounted per power panel. Each PU is capable of powering 4 terminals with adjuncts, for a total of 12 terminals per full MBPS. The 346 MBPS is intended to be installed in a closet and should be near the terminals (within 260 feet). All connections are modular and are made with cords and adapters at the 858A Adapter on the SIP. Terminals and Selector Consoles can be powered by the 346 MBPS.

Connectivity

System 25 requires 4-pair building wiring that conforms to AT&T Premises Distribution System (PDS) specifications. Various cords, cables, adapters, and connecting blocks are used to facilitate the connection of equipment and associated cable and wire.

Major points of connectivity include the following:

- The system cross-connect field located on a wall adjacent to the system cabinets. The field provides mounting space for the Trunk Access Equipment (TAE), Station Interconnect Panels (SIPS), and Emergency Transfer Units (ETUs).

  Refer to Section 5, “Environmental Requirements” for a typical System 25 layout including cross-connect field and associated equipment layout.

- 25-pair connectors located on the rear of each system cabinet.

- Modular jacks located at each work station provide modular connections for terminals and associated adjuncts and auxiliary equipment. These jacks are connected by building wiring to the SIP. Several wiring options are described below.

Wiring Options: There are three basic PECs under which building (station) wiring is ordered:

- PEC 2781-004 covers wiring done on an hourly rate.

- PEC 2782-004 covers flat rate wiring.

- PEC 2783-004 covers firm price quote.

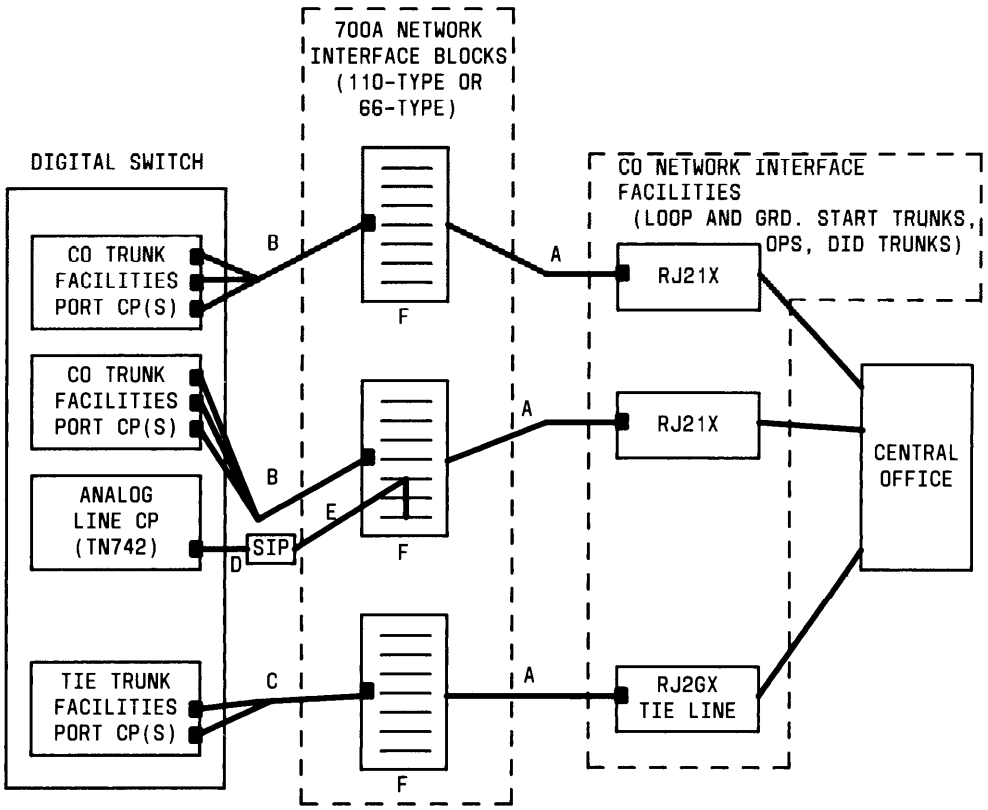
Consult the 2780 section of the Sales Manual for restrictions and requirements before ordering.

Trunk Access Equipment (TAE)

The TAE (Figure 4-23) provides for the connection of communications facilities such as Tie, OPS, Ground Start, Loop Start, and DID trunks to the trunk ports of the system. Up to three trunk CPs (except Tie Trunk CPs) can be connected to a 3-way splitter cable (PEC 2720-06X) that concentrates the CP interfaces into one 25-pair cable. Up to two Tie Trunk CPs can be connected to a 2-way splitter cable (PEC 2720-05X) that concentrates the CP interfaces into one 25-pair cable. Each splitter cable connects to an interface block at the TAE.

Cables are either cut down or plugged into the TAE blocks. The other end of the cables plug into the telephone company provided network interfaces (RJ21X or RJ2GX). Trunks and tie lines are cut down by the telephone company at the network interface.

700A or 157B Blocks are usually used for the TAE connections (furnished by the installer).



LEGEND:
- A - SINGLE-ENDED 25-PAIR CONNECTOR CABLE (A25D)*
- B - 3 TO 1 SPLITTER CONNECTORIZED CABLE - PEC 2720-06X
- C - 2 TO 1 SPLITTER CONNECTORIZED CABLE - PEC 2720-05X
- OPS - OFF-PREMISES STATION
- SIP - STATION INTERCONNECT PANEL*
- D - OCTOPUS CABLE - PEC 2720-05P
- E - INSIDE WIRE*
- F - TRUNK ACCESS EQUIPMENT (TAE) CONNECTOR BLOCK*
- * - FURNISHED BY INSTALLER

Figure 4-23. Trunk Access Equipment (TAE) Connections

Station Interconnect Panel (SIP)

The Station Interconnect Panel (SIP) provides for the connection of the terminals (voice and data), peripheral equipment, and some auxiliary equipment of the system to station port CPs. This equipment includes voice terminals, attendant consoles, data terminals, System Administration Terminal, Digital Tape Unit, and Call Accounting System. The SIP is made up of 617A Panels and associated adapters.

617A Panel

The 617A Panel (Figure 4-24) is a metal plate with keyslot holes on each side for mounting on a backboard. Each 617A Panel can hold eight Z210A1 or 858A Adapters, each of which can accommodate six connections to the port circuits in the cabinets. As many as five 617A Panels can be required for a maximum size system. The adapters snap into prepunched holes on the 617A Panels. (Reattached spacer buttons keep adapters from touching the metal panels.)

The cable rings located at the top of the 617A Panel route the building wiring cables to the adapters. Purse lock clips hold the building wiring cables in place. The white posts at the bottom of the 617A Panel guide the wiring from the 50A Fanning Strip to each column of adapters.

Preprinted boxes and numbers on the panel identify modular jacks for recordkeeping purposes. Letters are marked on the boxes at the top of each column by the installer. The letter (A-J) and the corresponding preprinted row number (1-24) identify the port jacks. For example, Al identifies the modular jack located in column A row 1.

Adapters

Adapters that mount on the panel connect the following:

- Building wire runs terminated in modular jacks, 25-pair connectors, or unterminated.
- Cables from the system cabinets terminated in modular jacks or 25-pair connectors.

The following adapters can be mounted on the 617A panel:

- Z210A—Six 4-pair modular jacks to six 4-pair modular jacks. One required per six voice terminals. Connects to building wiring terminated in modular plugs.
- 858A—Six 4-pair modular jacks to six 110-type cut-down blocks. One required per six voice terminals. Connects to unterminated building wiring.

The SIP equipment is furnished by the installer.

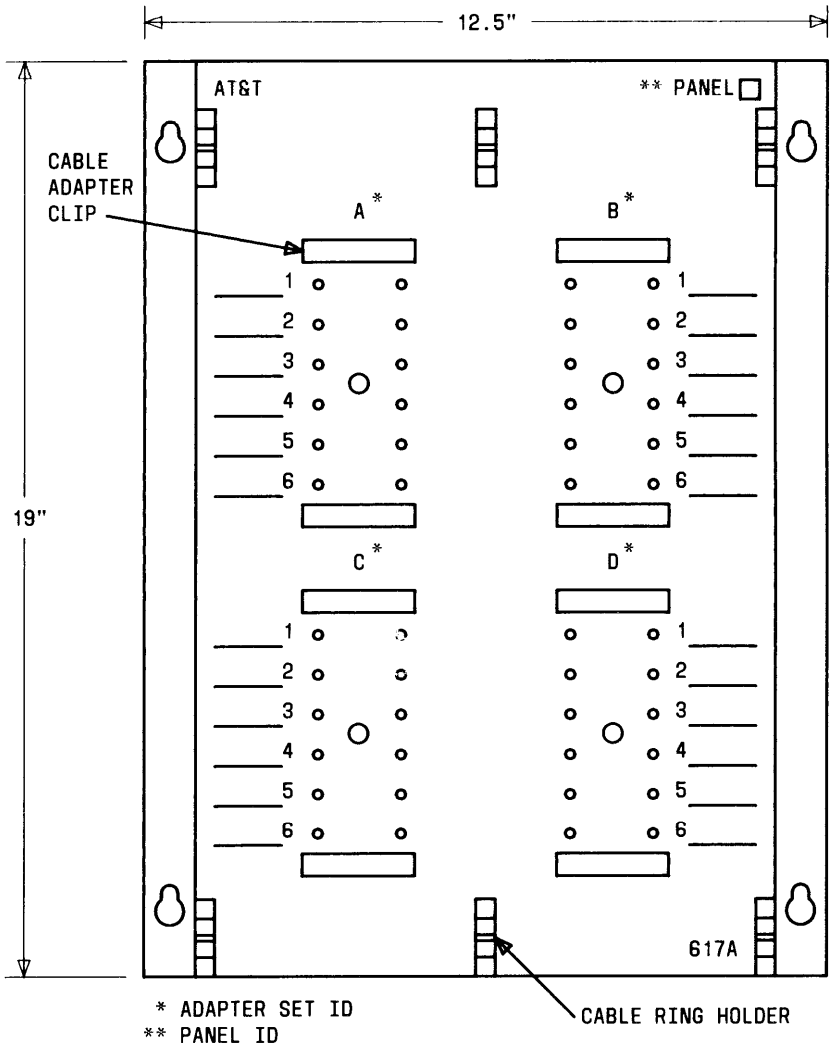


Figure 4-24. 617A Panel

Figure 4-25 shows voice terminal connections to the system cabinets via the SIP. Typically, voice terminals are plugged into modular wall jacks that provide a cut-down block for building wiring. At the SIP, 858A Adapters provide a cut-down point for 4-pair wire runs. An octopus cable (PEC 2720-05P) from a station CP provides 25-pair connectorized cabling to eight 4-pair modular jacks. Each jack is terminated on the SIP by an 858A Adapter. An octopus cable connects a maximum of eight voice terminals to a port CP.

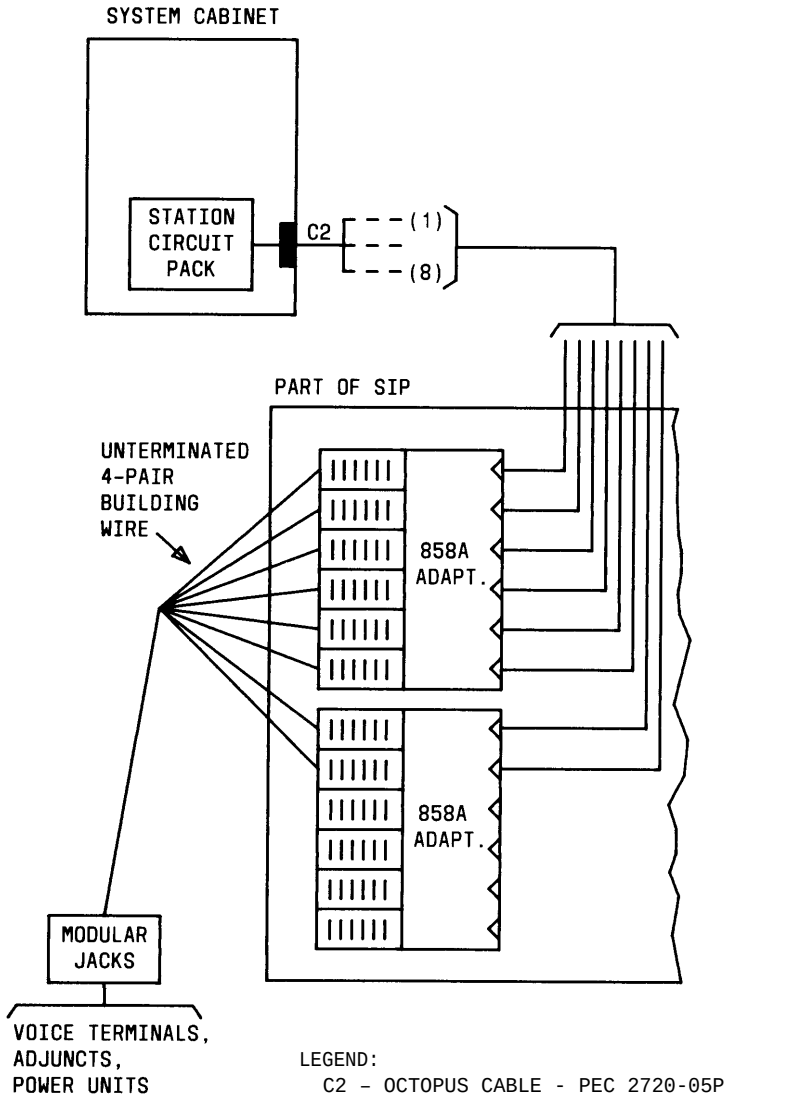


Figure 4-25. Typical SIP Connections

Connectivity Figures

Figures 4-26 through 4-36 provide connection information for various equipment. These figures have been included as an aid to understanding how equipment can be connected to System 25 and to indicate required connecting and supporting equipment. Other arrangements are possible; these figures can be useful in developing connecting arrangements for new or customer-provided equipment.

The PEC codes have been noted on the figures, as have indications of the source for obtaining non-PEC equipment (for example, from installer or furnished with other equipment). This information can be of use to Account Executives and Technical Consultants who are adding equipment to existing installations. For *new installations, the DOSS Configurator must be used to select equipment requirements.* For existing installations, you will need to determine what equipment is already installed. *You should not order equipment directly using the PECs in these figures.* The octopus cable (PEC 2720-05P), for example, supports eight terminals; you do not order one per terminal.

A list of related PECs, Apparatus, and Comcodes is provided in Section 7. Be sure to check the Sales Manual and/or DOSS before ordering since this information changes frequently.

Symbols Used in Figures: Modular jacks are shown by the triangle symbol. The 25-pair connectors are indicated by shaded blocks. Generally, only one leg of an octopus cable is shown. Unterminated wiring requiring cut down or other termination do not have symbol designations. The 103A Connecting Block is a typical modular wall jack that provides cut-down connections for building (station) wiring.

Voice Terminal and Adjunct Connections

Figures 4-26 through Figure 4-32 provide connection information for single-line and multiline voice terminals. The single-line terminals can be located on-premises, off-premises, or out-of-building. The MERLIN System multiline voice terminals can be used for out-of-building service but must be within 2000 feet of the system cabinets (local power is required beyond 1000 feet for in-building sets and for all out-of-building sets). Off-premises service is not available.

Diagrams for voice terminal adjunct connections are included with the specific feature descriptions in Section 2.

Attendant Console Connections

Diagrams for attendant console connections are included in the “Attendant Console” descriptions in Section 2.

Peripheral Equipment Connections

Diagrams for peripheral equipment connections are included with the specific feature descriptions in Section 2.

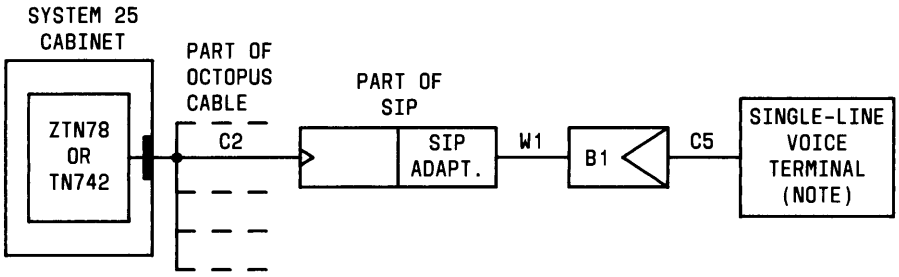
ADU Connections

Figures 4-33 and 4-34 provide connection information for data terminals and associated single-line or multiline voice terminals. The voice terminal and data terminal leads are separated at the SIP with a Y-adapter and are connected to their respective station ports. Figure 4-35 presents a typical Multiple Asynchronous Data Unit (MADU) connection.

Figure 4-36 shows local power connections for Z3A1, Z3A2, and Z3A4 ADUs.

Auxiliary Equipment Connections

Diagrams for auxiliary equipment connections are included with the specific feature descriptions in Section 2.



LEGEND:

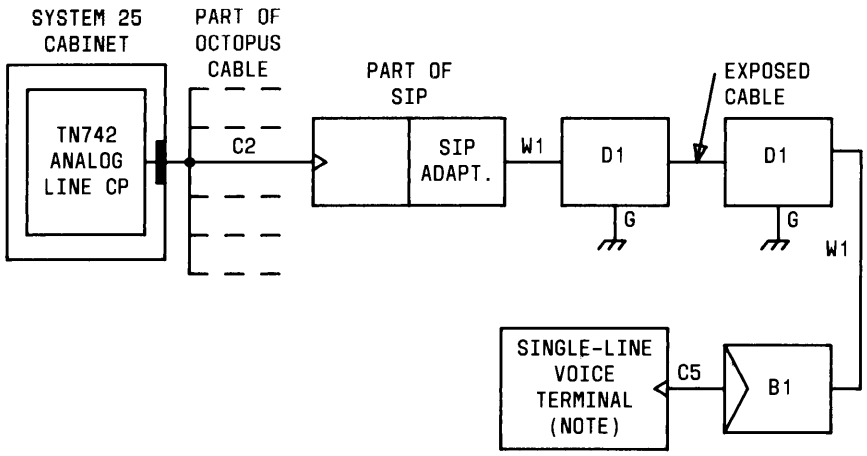
ZTN78 - TIP RING CP
TN742 - ANALOG LINE CP
B1 - TYPICAL-103A CONNECTING BLOCK*
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C5 - MODULAR CORD (D4BU-87) FURNISHED WITH SET
W1 - 4-PAIR INSIDE WIRING CABLE*

NOTE: RANGE LESS THAN 2000 FEET FROM SYSTEM CABINET, USE ZTN-78 CP. RANGE MORE THAN 2000 FEET BUT LESS THAN 1300 OHMS† (LOOP RESISTANCE) FROM SYSTEM CABINET, USE TN742 CP. FIVE SINGLE-LINE VOICE TERMINALS CAN BE BRIDGED WHEN USING A TN742 CP; HOWEVER, ONLY TWO MAY BE OFF-HOOK AT ONE TIME.

* FURNISHED BY INSTALLER

† - INCLUDES TELEPHONE/TERMINAL

Figure 4-26. On-Premises Single-Line Voice Terminal Connections

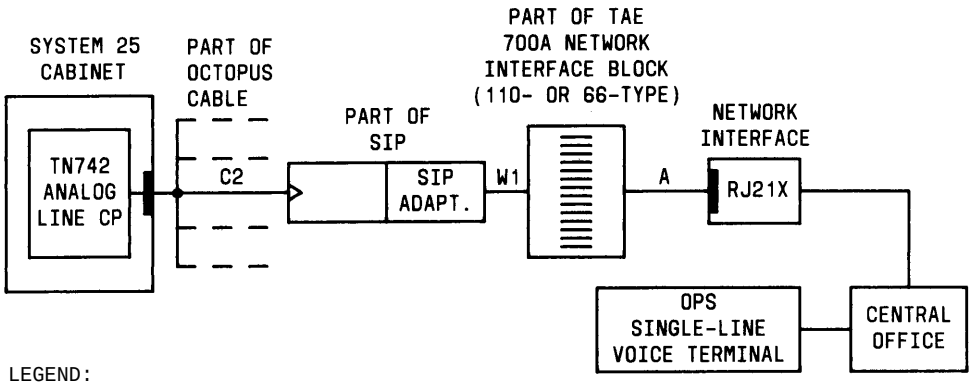


LEGEND:
- B1 - TYPICAL - 103A CONNECTING BLOCK*
- C2 - OCTOPUS CABLE (WP90780)
- C5 - MODULAR CORD (D4BU-87) - FURNISHED WITH SET
- D1 - STANDARD GAS TUBE/FUSE PROTECTION PER BSP 460-100-400*
- G - APPROVED BUILDING GROUND
- W1 - 4-PAIR INSIDE WIRING CABLE*

NOTE:
MAXIMUM LOOP RESISTANCE FROM SYSTEM CABINET <1300 OHMS†
FIVE SINGLE-LINE VOICE TERMINALS CAN BE BRIDGED; ONLY
TWO MAY BE OFF-HOOK AT ONE TIME.

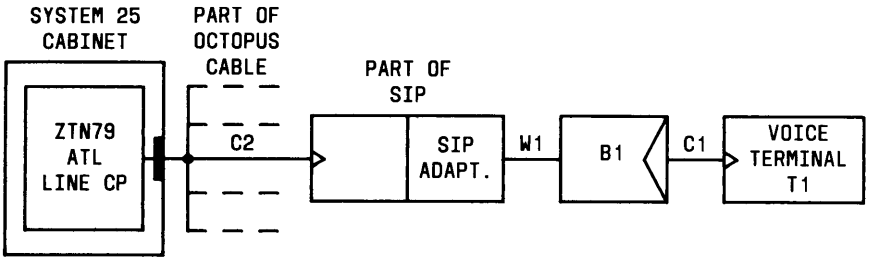
* - FURNISHED BY INSTALLER
† - INCLUDES VOICE TERMINAL

Figure 4-27. Out-of-Building Single-Line Voice Terminal Connections



LEGEND:
A - SINGLE-ENDED 25 PAIR CABLE (A25D)*
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
W1 - 4 PAIR INSIDE WIRING CABLE*
* - FURNISHED BY INSTALLER

Figure 4-28. Off-Premises Station Single-Line Voice Terminal Connections



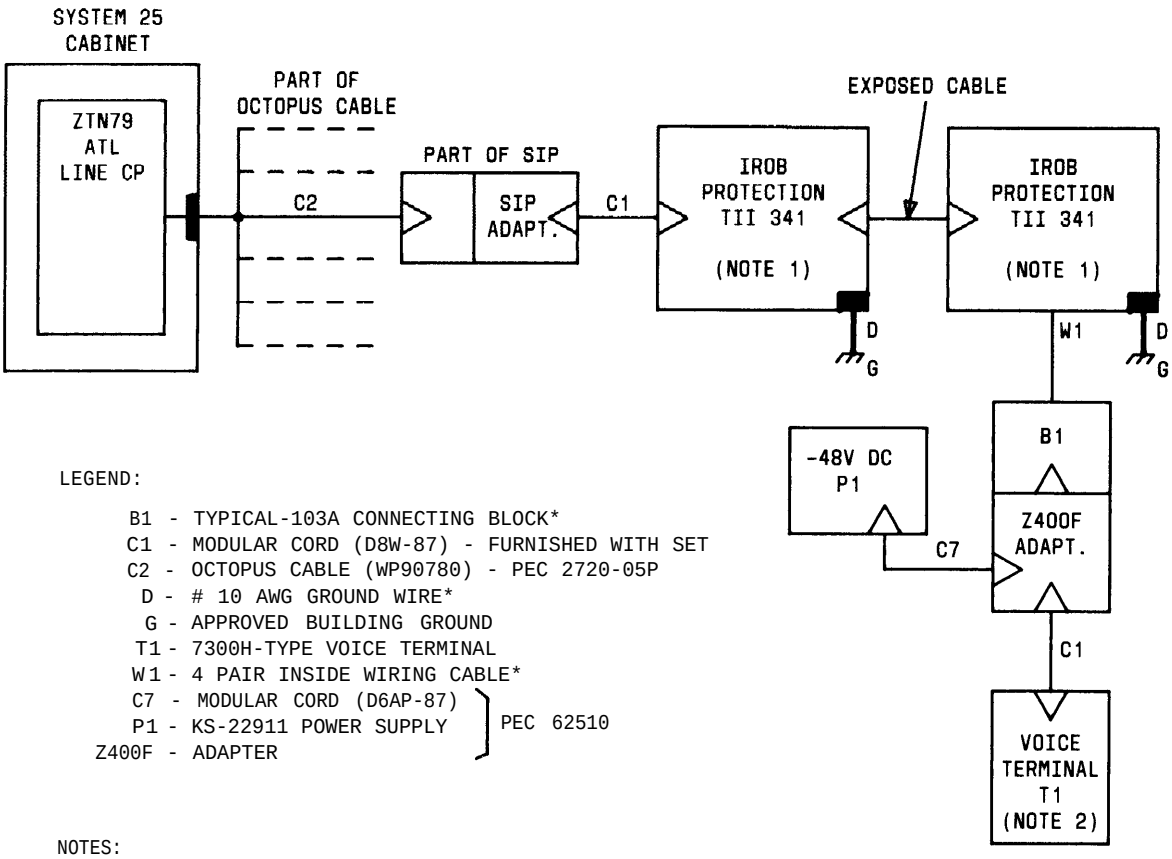
LEGEND:

B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
T1 - 7300H TYPE VOICE TERMINAL
W1 - 4 PAIR INSIDE WIRING CABLE*

* - FURNISHED BY INSTALLER

NOTE: RANGE WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED BEYOND 1000 FEET).

Figure 4-29. On-Premises 7300H Series Multiline Voice Terminal Connections



LEGEND:

B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
D - # 10 AWG GROUND WIRE*
G - APPROVED BUILDING GROUND
T1 - 7300H-TYPE VOICE TERMINAL
W1 - 4 PAIR INSIDE WIRING CABLE*
C7 - MODULAR CORD (D6AP-87)
P1 - KS-22911 POWER SUPPLY } PEC 62510
Z400F - ADAPTER

NOTES:

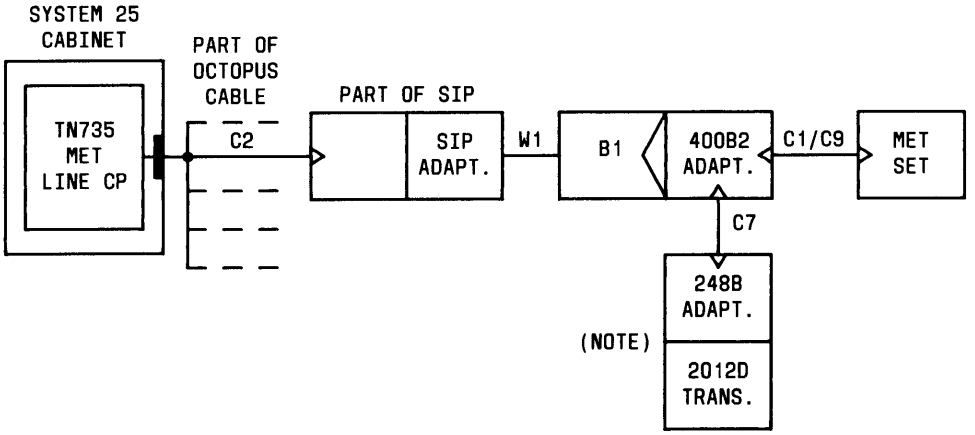
1. IROB (IN-RANGE-OUT-OF-BUILDING) PROTECTION: 2 TII 341 MODELS REQUIRED (SYSTEM-STATION ENDS) FOR PRIMARY/SECONDARY PROTECTORS.

   CAUTION: DO NOT CONNECT IROB POWER LEADS.

2. RANGE: WITHIN 2000 FEET OF SYSTEM CABINET.

* FURNISHED BY INSTALLER

Figure 4-30. Out-of-Building 7300H Series Multiline Voice Terminal Connections



LEGEND:

B1 - TYPICAL-103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C9 - ADAPTER CORD (ZD8AJ-87) - PEC 2750-A17
MET SETS - 2991C05, 2993C04, AND 2991D05 10-BUTTON MET SETS- USE C9 AND C1
MET SETS - 7203M 12-BUTTON MET SET - USES Cl ONLY
W1 - 4-PAIR INSIDE WIRING CABLE*

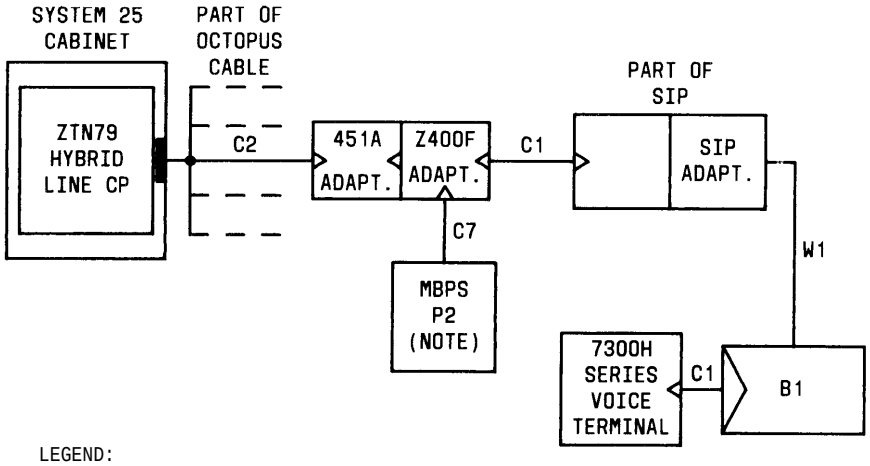
248B ADAPTER - MODULARIZES 2012D TRANSFORMER
400B2 ADAPTER - POWER ADAPTER
2012D TRANSFORMER - 15-18V AC SOURCE
C7 - MODULAR CORD (D6AP-87)

} PEC 21691

NOTE: ONLY MET SET WITH BUILT-IN SPEAKERPHONE (2993C04) REQUIRES TRANSFORMER AND ADAPTERS. OTHERWISE, PLUG C1 INTO B1 DIRECTLY.

* - FURNISHED BY INSTALLER

Figure 4-31. MET Set Connections

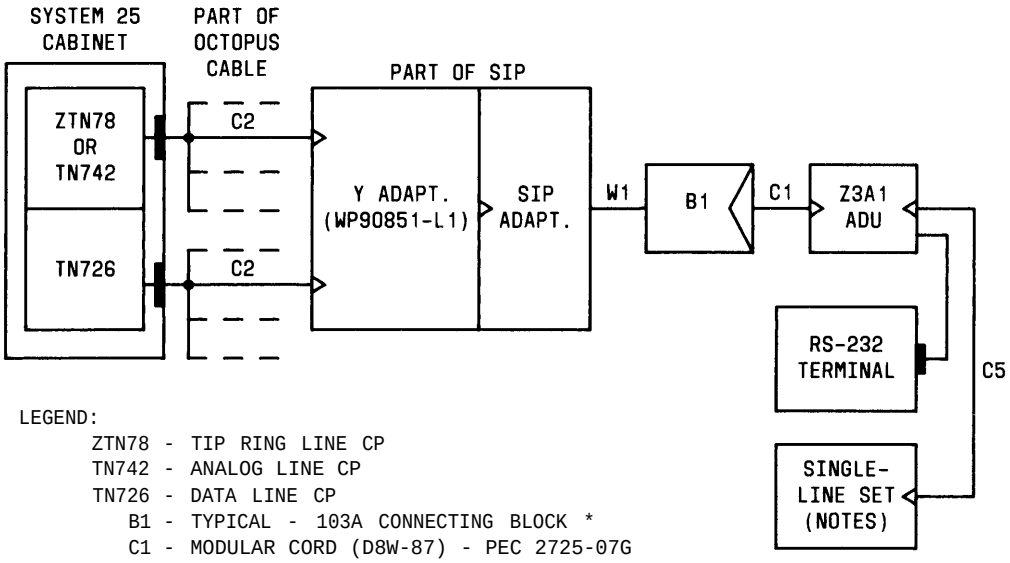


LEGEND:

B1 - TYPICAL – 103A CONNECTING BLOCK*
C1 - MODULAR CORD (D8W-87)
C2 - OCTOPUS CABLE (WP90780)
C7 - MODULAR CORD (D6AP-87)
P2 - MODULAR BULK POWER SUPPLY, CONSISTING OF:
POWER UNIT (346A) - PEC 31760
POWER PANEL (346A1) - PEC 31761
W1 - 4-PAIR INSIDE WIRING CABLE'
* - FURNISHED BY INSTALLER

NOTE: PEC 62510 CAN ALSO BE USED TO SUPPLY LOCAL POWER.

Figure 4-32. Stand-Alone Remotely Powered Voice Terminal Connections



LEGEND:

ZTN78 - TIP RING LINE CP
TN742 - ANALOG LINE CP
TN726 - DATA LINE CP
B1 - TYPICAL - 103A CONNECTING BLOCK *
C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
C5 - MODULAR CORD (D4BU-87) - FURNISHED WITH SET
W1 - 4-PAIR INSIDE WIRING CABLE *
WP90851-L1 - MODULAR Y ADAPTER - PEC 2750-T05
Z3A1 ADU - E/W 3-FT PLUG-ENDED EIA CORD - PEC 2169-001

NOTES :

1. IF RANGE IS GREATER THAN 2000 FT FROM SYSTEM CABINET, TERMINAL DATA RATE (SPEED) WILL BE LIMITED (SEE “CABLE DISTANCE LIMITATIONS” IN SECTION 5)
2. IF RANGE IS LESS THAN 2000 FEET FROM SYSTEM CABINET USE ZTN-78. IF RANGE IS MORE THAN 2000 FEET BUT LESS THAN 1300 OHM† (LOOP RESISTANCE) FROM CABINET, USE TN-742

* - FURNISHED BY INSTALLER
† - INCLUDES TELEPHONE/TERMINAL

Figure 4-33. Typical ADU Connections—Supporting Data Terminal and Single-Line Voice Terminal

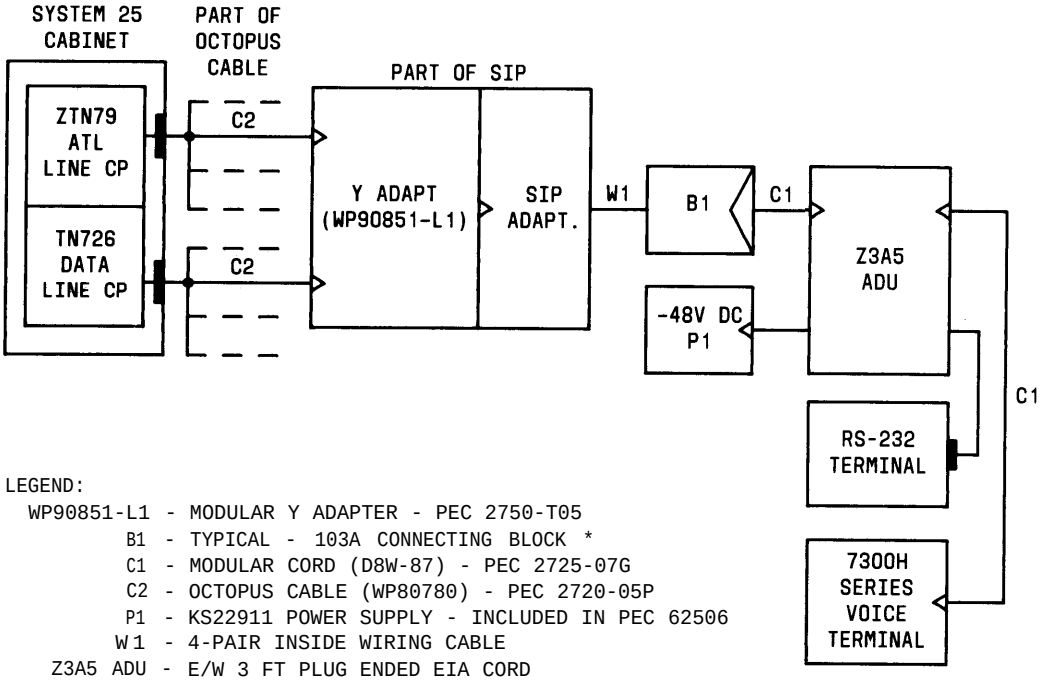
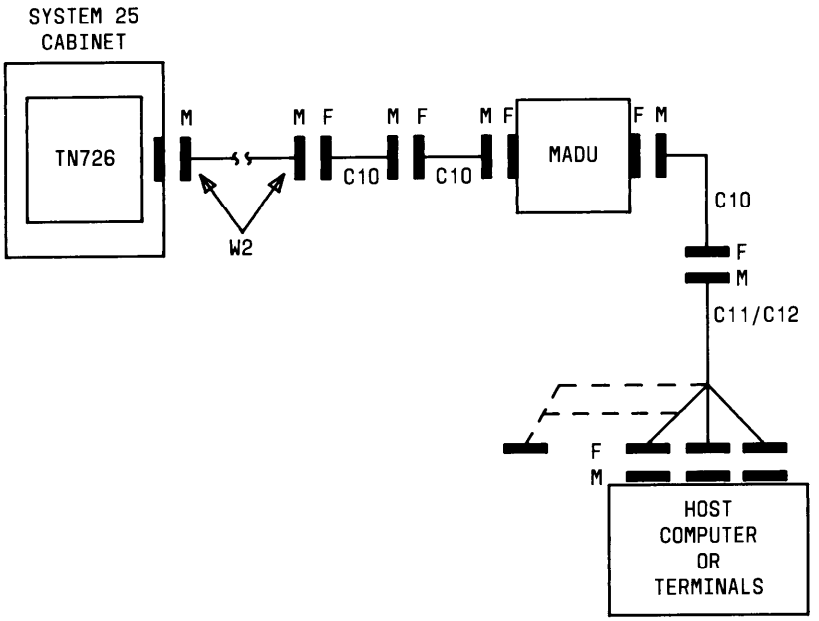
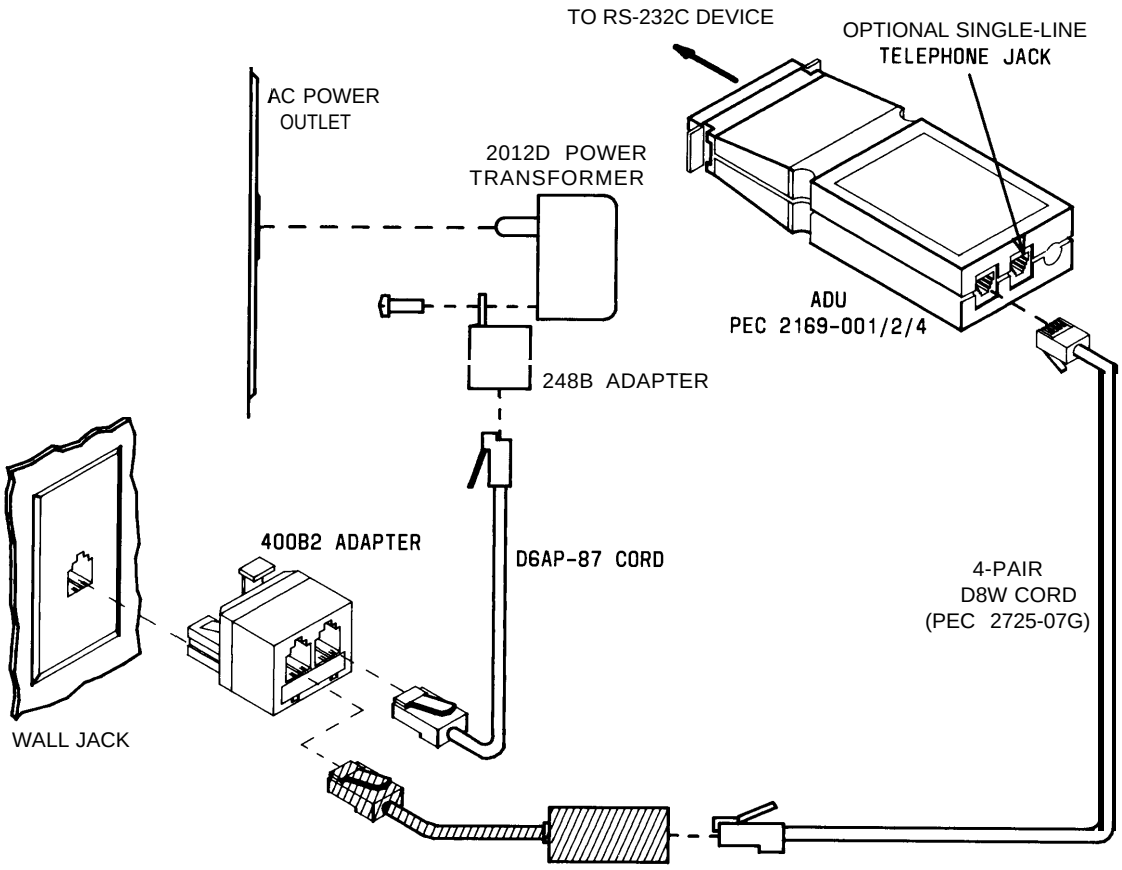


Figure 4-34. Typical ADU Connections—Supporting Data Terminal and 7300H Series Multilane Voice Terminal



LEGEND:
TN726 – DATA LINE CP
MADU – MULTIPLE ASYNCHRONOUS DATA UNIT (PEC 2169-005)
SINGLE UNIT ASSEMBLY (8 ADU PORTS)
W2 – BUILDING WIRING (25-PAIR CABLE)
C10 – 25-PAIR CENTERFEED-TO-ENDFEED CABLE (PEC 2724-78B)
(ALWAYS REQUIRED)
C11– M48C OCTOPUS CABLE (PEC 2724-29G) (7-FOOT CORD WITH
EIGHT 6-INCH ARMS FOR “DTE” HOST INTERFACE
C12 – M48G OCTOPUS CABLE (PEC 2724-98G) (7-FOOT CORD WITH
EIGHT 6-INCH ARMS FOR “DCE” HOST INTERFACE

Figure 4-35. Typical MADU Connections



NOTE :
PEC 21691 INCLUDES 2012D TRANSFORMER, 248B AND 400B2 ADAPTERS AND D6AP CORD.

Figure 4-36. Z3A1/2/4 ADU Local Power Connections

TECHNICAL SPECIFICATIONS

Hardware and Software Parameters 5-2

Unit Loads 5-6

Cable Distance Limitations 5-7

Call Progress Tones 5-10

Indicator Lamp Signals 5-11

Port Specifications 5-12

Recommended Central Office Trunk Facilities 5-18

Analog Transmission Characteristics 5-19

Figures

Figure 5-1	Single-Line Voice Terminal Allowable Cable Distances	5-7
Figure 5-2	Multiline Voice Terminal Allowable Cable Distances	5-8
Figure 5-3	Asynchronous Data Unit Allowable Cable Distances (in-Building Only)	5-9

TECHNICAL SPECIFICATIONS

This section provides information on the technical characteristics and capacities of the system. Some items covered here are discussed elsewhere in the manual but are repeated here for ease of reference.

Technical specifications are provided for:

- Hardware and Software Parameters
- Unit Loads
- Cable Distance Limitations
- Call Progress Tones
- Indicator Lamp Signals
- Port Specifications
- Recommended Central Office Facilities (Trunks)
- Analog Transmission Characteristics.

Hardware and Software Parameters

The following is a listing of maximums for hardware and software parameters.

ITEM	TOTAL
Attendant Consoles	
Direct Trunk or Switched Loop	2
Direct Extension Selector	2
Automatic Route Selection (ARS):	
Patterns	8
Subpatterns Per Pattern	2
Routes Per Subpattern	3
Facility Restriction Levels	4
Cabinets	3
Circuit Packs:	
Common Control and Service Circuit	2
Circuit Pack Slots Per Cabinet	12
Conference/Bridging Members	5
Coverage	
Individual Covered Stations	No limit
Coverage Groups—Standard Group Coverage	32
Coverage Groups—DGC Group Coverage	32
Receivers per covered station or standard group	8
Dial Codes	600
Personal Dial Codes (PDCs)	200
Floating Personal Dial Codes (FPDCs)	300
Data Dial Codes (DDCs)	104
Direct Group Calling (DGC)	
Groups	32
Members per group	20

Hardware and Software Parameters (Contd)

ITEM	TOTAL
Emergency Transfer Units (ETUs) Voice Terminals per ETU	4 5
Modem Pool Circuit Packs per cabinet Conversion Resources per circuit pack	2 2
Paging Zones	3
Parked Calls (System) Per Voice Terminal Attendant Selector Console	24 1 8
Pickup Groups Members per group	 16 16
Trunk Groups	16
System Delay Announcements: Direct Group Calling Delay Announcement Directed Night Service Delay Announcement	 1 2
Account Code Digits SMDR or Call Accounting System Models 100, 200, 300, or 500)	15
Speed Dialing Numbers System Speed Dialing Numbers (#100-#189) Personal Speed Dialing Numbers (#20-#39), per station Repertory Dialing *digits* plus Speed Dialing *digits*	 90 20 34100

Hardware and Software Parameters (Contd)

ITEM	TOTAL
System Administration Terminal .	1
Toll Call Allowed Lists Total Entries (all lists)	4 64
Virtual Facilities	10
Traffic Data • Simultaneous 2-Party Conversations • Call Capacity —CCS/Hour —Busy Hour Call Capacity Reliability • Mean Time Between Outages (MTBO) Power Consumption • Per Cabinet, Maximum • Thermal Dissipation	 115 4140 2500 4 Years 500 Watts 1700BTU/Hour
Total Ports - also includes trunk and station ports [Software Limits; hardware maximum = 36 CPs] System Resources • One Service Circuit CP (includes 4 TT Receivers) (8 ports allocated/CP) • Two Touch-Tone Receiver CPs (4 TT Receivers/CP) (8 ports allocated/CP) • Pooled Modem (Max 6 CPs, 4 ports allocated/CP) (two modems per CP)	256 8 16 24

Hardware and Software Parameters (Contd)

ITEM	TOTAL
Trunks	
• Trunk Ports	104
—Tie Trunks	32
• Auxiliary Trunk Ports	
—Paging Access	3
—Dictation Access	8
Station Ports	240
• Data Ports	104
• Voice Ports	200
—Single-Line Voice Terminals	200
—Multiline Voice Terminals	144
Attendant Consoles	2
Selector Consoles	2
22- or 34-Button Sets (nonattendant)	96
Non 22- or 34-Button Sets	142

Unit Loads

A cabinet can supply no more than 80 unit loads of 48 volt power (a unit load is defined as 44 mA). Unit loading is determined by the terminal connected to the port circuits. The following table lists unit loads for various terminals.

UNIT LOADS (Note)		
EQUIPMENT	**CIRCUIT PACK**	**UNIT LOAD PER PORT**
2500 Voice Terminals 7100 Voice Terminals	ZTN78	0.5
	TN742	1.0
MET Sets (10 Btn.) MET Set (12 Btn.)	TN735 TN735	1.0 2.0
5-Btn. (7302H01D) 10-Btn. (7303H01D) 34-Btn. (7305H01D) HFAI (7309H01B)	ZTN79	1.0 1.0 1.0 1.0
BIS (7305H03B) BIS w/display (7305H04C)	ZTN79 ZTN79	1.50 2.00
34-Btn. Deluxe (7305H02D)	ZTN79	2.0
BIS-10 (7313H01A) BIS-22 (7314H01A) BIS-34 (7316H01A) BIS-34D (7317H01A)	ZTN79	1.2 2.0 2.0 2.0
S102A Speakerphone	ZTN79	1.0
502A/B Headset Adapter	ZTN79	0.8
Asynchronous Data Units	TN726	0.0
Tie Trunks	TN760B	2.0
DID Trunks	TN753	0.5
Emergency Transfer Unit	ZTN129	2 per ETU
Pooled Modem	TN758	0.0'
DXS Selector Console	Locally Powered	

* Zero Unit Loads but maximum of two TN758s/cabinet allowed.

Note: Equipment not listed above (i.e., TN763, ZTN76, ZTN77) does not affect unit loading. Any voice terminal/adjunct combination requiring more than 2 Unit Loads must be locally powered. When a voice terminal is locally powered, it places no unit load on the cabinet. Specifically, a 34-Button Deluxe voice terminal equipped for speakerphone operation requires auxiliary power. In addition, any 22- or 34-Button Deluxe, BIS, or BIS with Display voice terminal equipped for headset operation requires auxiliary power.

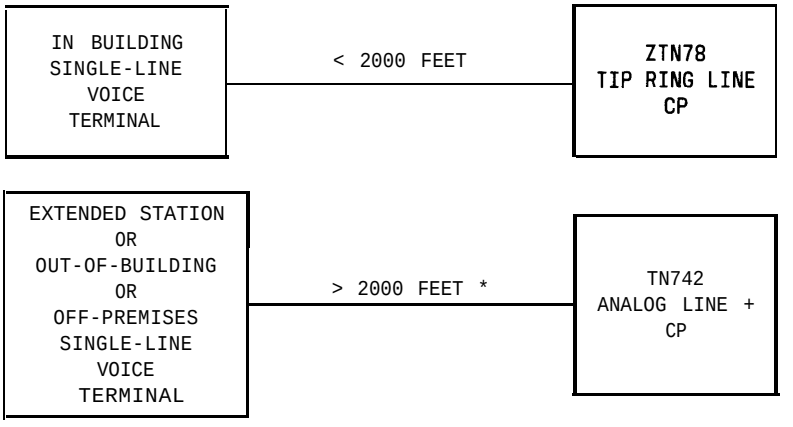
Cable Distance Limitations

This subsection provides allowable cabling distances for the following devices:

- Single-Line Voice Terminals (Figure 5-1)
- Multiline Voice Terminals (Figure 5-2)
- Data Terminals (RS-232) Connected to Asynchronous Data Units (ADUs) (Figure 5-3).

Single-Line Voice Terminals

SUPPORTING CIRCUIT PACK	24-GAUGE WIRE (0.5106 mm)	
	FEET	METERS
ZTN78	2,000	610
TN742 (2500 Series)	17,500	7,320
TN742 (Model 7101A)	15,000	4,500



* UP TO 24,000 FEET FOR OUT-OF-BUILDING. TIP/RING LOOP RESISTANCE FROM SYSTEM CABINETS (INCLUDING VOICE TERMINAL) MUST NOT EXCEED 1300 OHMS.

\+ FIVE SINGLE-LINE VOICE TERMINALS CAN BE BRIDGED WHEN USING THE TN-742, HOWEVER, ONLY TWO MAY BE OFF-HOOK AT ONE TIME.

Figure 5-1. Single-Line Voice Terminal Allowable Cable Distances

Multiline Voice Terminals

SUPPORTING CIRCUIT PACK	24-GAUGE WIRE (0.5106 mm)	
	FEET	METERS
ZTN79 (7300H Series—in-building or out-of-building, no off-premises)	2,000 (Note)	610
TN735 (in-building MET Sets only)	1,000	305

Note: Requires local power (PEC 62510) beyond 1,000 feet or whenever IROBS are used.

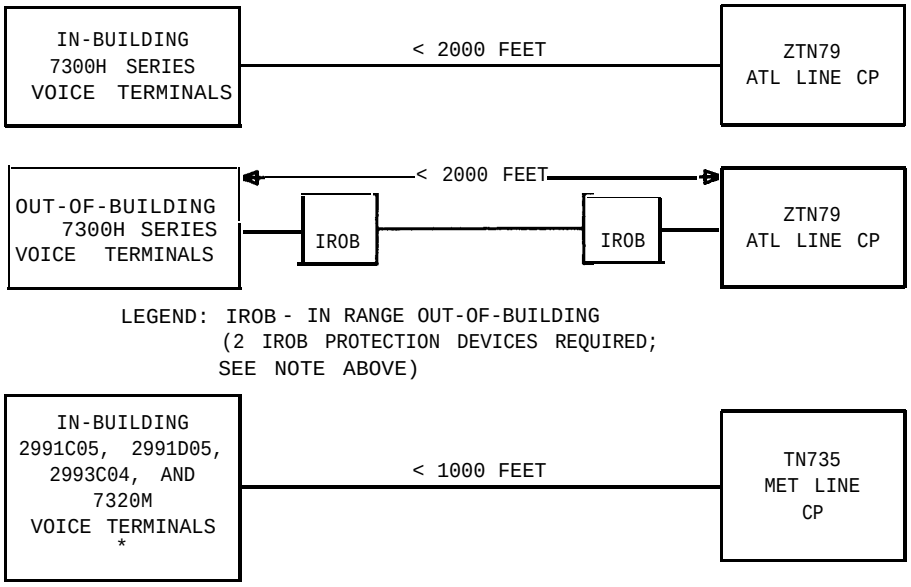


Figure 5-2. Multiline Voice Terminal Allowable Cable Distances

Data Terminals (RS-232) Connected To Asynchronous Data Units

DATA RATE	24-GAUGE WIRE (0.5106 mm)	
	FEET	METERS
300 bps	40,000	12,200
1,200 bps	20,000	6,096
2,000 bps	12,000	3,657
4,800 bps	7,000	2,133
9600 bps	5,000	1,524
19,200 bps	2,000	610

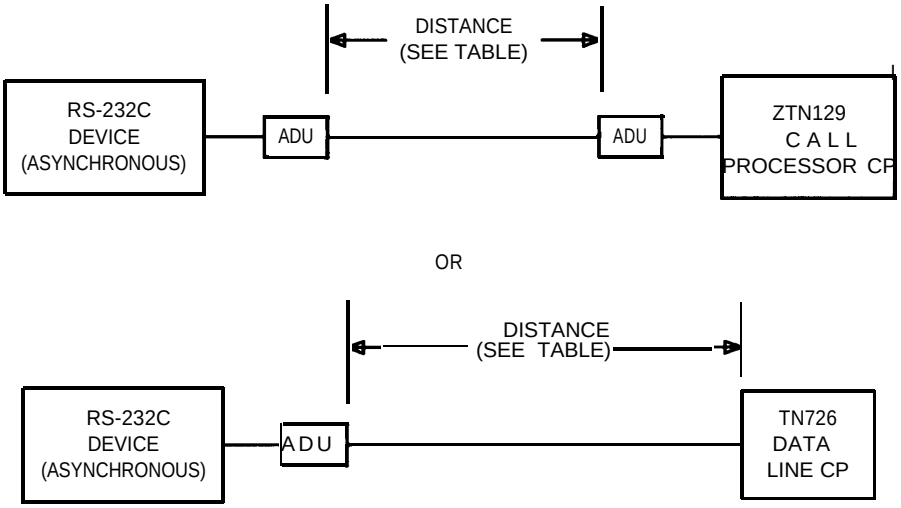


Figure 5-3. Asynchronous Data Unit Allowable Cable Distances (in-Building Only)

Call Progress Tones

The following call progress tones are generated by the system:

TONE	FREQUENCY	PATTERN (In Milliseconds)
Busy	480 Hz + 620 Hz	500 on, 500 off; repeated
Call Waiting Notify	440 Hz	200 on; not repeated
Confirmation	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on followed by silence
Dequeuing	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on followed by silence
Dial	350 Hz + 440 Hz	Continuous
Queuing	440 Hz	Five 50 ms tones, 50 ms apart, not repeated
Recall Dial	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on, 100 off, followed by continuous tone
Reorder	480 Hz + 620 Hz	250 on, 250 off; repeated
Ringback, Normal	440 Hz + 480 Hz	1200 on, 4000 off; repeated
Ringback, Special (Call Waiting)	440 Hz + 480 Hz; 440 Hz	1000 on (440 Hz + 480 Hz), 200 on (440 Hz), 2800 off; repeated

Indicator Lamp Signals

The following lamp signals are provided at multiline voice terminal line appearances:

LAMP SIGNAL	PATTERN (In Milliseconds)	MEANING
Dark	Off	Inactive
Lighted	On	Active
Flashing	500 on, 500 off; repeated	Ringing
Broken Fluttering	50 on, 50 off; repeated, (gated on/off every 500 ms)	Transfer/Conference in progress
Wink	350 on, 50 off; repeated	Hold

Port Specifications

The following tables provide interface specifications for System 25 line and trunk port circuits:

DATA TERMINAL PORTS (Note)		
STATION TYPE	**CIRCUIT PACK**	**DATA TERMINAL SPECIFICATIONS**
EIA RS-232 Device Via ADU	Data Line (TN726)	RS-232 device must furnish signals on ADU pins 2 (TD) and 20 (DTR) and ground on either pin 1 or 7. The ADU furnishes signals on pins 3 (RD) and 8 (CD). The CD signal is also tied to pins 5 (CTS) and 6 (DSR). Some data terminals may require auxiliary power when used with a Z3A1, Z3A2, or Z3A4 ADU (the Z3A5 always requires local power). ADUs require 7 volts on pin 20 (DTR) to operate properly. The following table lists data terminals that have been tested and are known to operate properly *without* auxiliary power.

Note: Refer to Cable Distance Limitations for supported data rates.

Port Specifications (Contd)

DATA TERMINALS (Do Not Require Local Power)		
DATA TERMINAL	**REQUIRES Z3A1 or Z3A2 ADU**	**REQUIRES Z3A4 ADU**
AT&T		
4410	x	
4415	x	
510A	x	
ADMs		
3A	x	
31	x	
ADDS		
Viewpoint *	x	
ConCept		
HDS 108		x
Datamedia		
Elite 1521	x	
Hazeltine		
1510	x	
Hewlett Packard		
2621A		x
2623A		x
2640		x
2645		x
2645A		x
Teletype		
BLIT/1 (68000 based)	x	
5620 (MAC-80 based)	x	
5420	x	

* Requires Originate/Disconnect Switch.

Port Specifications (Contd)

VOICE TERMINAL PORTS		
STATION TYPE	**CIRCUIT PACK**	**SPECIFICATIONS**
Tip and Ring Single-Line Sets (Analog)	Tip Ring Line (ZTN78)	• 1-Pair Interface (Tip and Ring) • Analog signals modulated over DC loop • Loop Voltage: 24 V dc • Signaling: Dual Tone Multifrequency (DTMF) or Dial Pulse • REN (max.): 1.2 • DC Current (max.): 35 mA • Loop Range (24 AWG): 2,000 feet • In-building service only
Tip and Ring Single-Line Sets (Analog)	Analog Line (TN742)	• 1-Pair Interface (Tip and Ring) • Analog signals modulated over DC loop • Loop Voltage: 48 V dc • Signaling: Dual Tone Multifrequency (DTMF) or Dial Pulse • REN (max.): 5.0 • DC Current (max.): 40 mA • Loop Range (24 AWG): 17,500 feet • Supports Out-of-Building, Extended (greater than 2000 feet), Off-Premises, and Bridged Station services (maximum of five bridged stations, and two off-hook simultaneously).

Port Specifications (Contd)

VOICE TERMINAL PORTS (Contd)		
STATION TYPE	**CIRCUIT PACK**	**SPECIFICATIONS**
MET Sets (Hybrid)	MET Line (TN735)	• 3-Pair Interface 1-Voice pair 2-Control pairs • Analog Voice, Digital Control/Signaling • Power: Phantom Power Over Data Pairs • Bipolar Signaling With 0 V dc Offset • 1 MHz Nominal Signaling Rate • Loop Range: 1000 feet (in-Building service only)
MERLIN® System Sets (7300H Series) (Hybrid)	ATL Line (ZTN79)	• 3-Pair Interface 1-Voice pair 1-Control pair 1-Power pair • Analog Voice, Digital Control/Signaling • Bipolar non-return to zero line-coding • 40 kHz Nominal signaling rate • Loop Range: 1000 feet • In-Building and In-Range Out-of-Building (IROB) services only

Port Specifications (Contd)

TRUNK PORTS		
TRUNK TYPE	**CIRCUIT PACK**	**SPECIFICATIONS**
Auxiliary Trunk	TN763	• Capacity: 4 Circuits • 3-pair Interface: Voice (T, R), Signaling (S, S1), Status (SZ, SZ1)
Direct Inward Dialing (DID) Trunk	TN753	• Capacity: 8 Circuits • 2-Wire (600 Ohm Fixed Impedance) Transmission • Signaling: Wink Start, Delay Dial, or Immediate Dial. Accepts Dial Pulse Signals only • Incoming Service only
Tie Trunk	TN760B	• Capacity: 4 Circuits • Supports Type I E&M, Type I Compatible E&M or Type V Simplex Signaling • 4-Wire Transmission • 3-Pair Interface Transmit Receive Signaling (E&M)

Port Specifications (Contd)

TRUNK PORTS		
TRUNK TYPE	**CIRCUIT PACK**	**SPECIFICATIONS**
Ground Start Trunk	ZTN76	• Capacity: 8 Circuits • 2-Wire (600 Ohms or RC Balance) Transmission • Network Signaling: Ground Start • Two-way or Incoming-only Service
Loop Start Trunk	ZTN77	• Capacity: 8 Circuits • 2-Wire (600 Ohms or RC Balance) Transmission • Network Signaling: Loop Start • Two-way or Incoming-only Service

Recommended Central Office Trunk Facilities

The following table provides recommendations for CO trunks based on the number of voice terminals in the system and the calling traffic.

VOICE TERMINALS	CALLING TRAFFIC		
	LIGHT TRAFFIC	MEDIUM TRAFFIC	HEAVY TRAFFIC
20	3/3	4/4	5/5
25	3/4	5/5	6/6
30	4/4	6/5	8/7
40	4/4	6/6	9/8
50	5/4	7/6	10/8
60	5/4	8/7	11/9
70	5/5	8/7	1 2/1 o
80	6/5	9/7	1 2/1 o
100	6/5	10/8	14/12
120	7/6	11/9	16/13
140	7/6	1 2/1 o	17/14
160	8/7	13/10	19/15
180	8/7	13/11	20/1 6
200	9/8	14/12	22/1 8

Notes:

1. For systems with both one-way and two-way trunks, the first number listed under “Calling Traffic” is the number of two-way trunks required, the second number is the number of one-way trunks required.

2. For systems with just two-way trunks, add the two numbers listed under Calling Traffic to determine the number of trunks required.

Analog Transmission Characteristics

Frequency Response:
(Station-To-Station or Station-To-CO Trunk, relative to loss at 1 kHz)

FREQUENCY	LOSS
60 Hz	>20 dB
200 Hz	<5 dB
300-3000 Hz	<1 dB
3200 Hz	<1.5 dB
3400 Hz	<3 dB

Insertion Loss:

CONNECTION TYPE	LOSS
Standard Station to Standard Station	6 dB
Standard Station to Extended/Off-Premises Station	3 dB
Extended/Off-Prem Station to Extended/Off-Prem Station	0 dB
Station-to-Trunk	0 dB
Trunk-to-Trunk	0 dB

Overload Level: +3 dBm0

Crosstalk: <-70 dB

Intermodulation Distortion:

FOUR TONE METHOD	
2nd Order Tone Products	>45 dB
3rd Order Tone Products	>53 dB

Analog Transmission Characteristics (Contd)

Quantization Distortion:

SIGNAL LEVEL	DISTORTION LEVEL
+2 to -30 dBm0	35 dB
-40 dBm0	29 dB
-45 dBm0	25 dB

Sampling Rate: 8 kHz

Terminating Impedance: 600 ohms

Trunk Balance Impedance: 600 ohms or Complex Z (selectable)

Echo Return Loss:

The echo return loss of the switching equipment is infinite. The echo return loss of the station equipment can be engineered for greater than 18 dB over the range of 500 Hz to 2500 Hz.

Loop Resistance:

- TN742—Loop resistance of up to 1300 ohms, including the station
- ZTN78—Loop resistance of up to 100 ohms **not** including the station (2000 feet with No. 24 AWG.)

Connection Bandwidth: 64 Kbits

Steady State Noise Level:

The steady state noise level presented to any busy path does not exceed 23 dBrnc during the busy hour.

Impulse Noise:

The impulse noise is 0 count (hits) in 5 minutes at +55 dBrnc during the busy hour.

Analog Transmission Characteristics (Contd)

Single Frequency Return Loss (Talking State):

Station to station—exceeds 12 dB
Station to 4-wire trunk connection—exceeds 14 dB
Station to 2-wire trunk connection—exceeds 12 dB

Peak Noise Level:

Analog to analog—20 dBrnc
Analog to digital—19 dBrnc
Digital to analog—13 dBrnc

ENVIRONMENTAL REQUIREMENTS

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Table Top Space	6-4
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Figures

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ENVIRONMENTAL REQUIREMENTS

This section provides information on floor and wall space requirements for System 25 cabinets and associated peripheral equipment. Also included are specifications for temperature, humidity, air purity, lighting, electrical noise (RFI) suppression, power, grounding, and lightning protection.

Floor Plans And Layouts

Floor plan arrangements will vary depending on the available equipment area and anticipated system growth. A typical floor plan is shown in Figure 6-1.

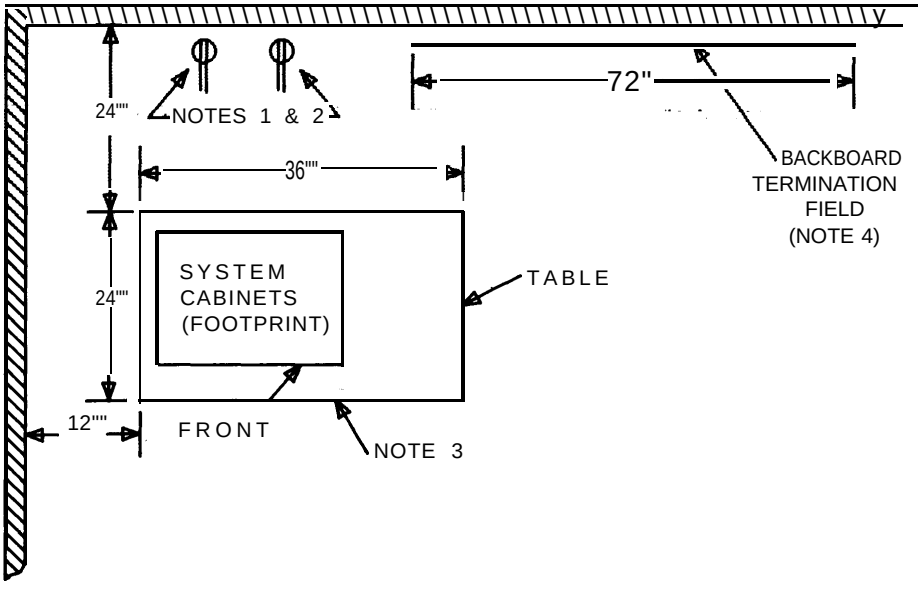
The floor must be tiled or suitably sealed, level, and free from vibration. Allow for a minimum unobstructed clearance of 7 feet above the floor throughout the equipment area.

Do not locate the equipment in areas:

- Where it might be subjected to excessive vibrations or disturbed by moving equipment such as hand trucks and transporters.
- Where noise levels may exceed 90 dB.
- Susceptible to flooding.

Maintain clear access to the equipment area for both installation and maintenance purposes.

The wall behind the system cabinet must be clear of all objects (pictures, shelves, or windows) that might interfere with system installation. The entire area behind the cabinet and to the side as shown on Figure 6-2 must be reserved for the cross-connect field and cable access. Also, room for system growth should be considered.



NOTES :

1. 115V AC, 60 Hz, 15 AMP OUTLETS (HUBBELL 5262 OR EQUIVALENT) MUST BE LOCATED WITHIN SIX FEET (1.8m) OF SYSTEM CABINETS.
2. MULTIPLE CABINET SYSTEMS REQUIRE TWO QUAD OUTLETS, SINGLE CABINET SYSTEMS REQUIRE ONE QUAD OUTLET.
3. ALLOW AT LEAST 24 INCHES OF SPACE IN FRONT OF CABINETS. TABLE MUST BE ABLE TO SUPPORT 250 POUNDS.
4. BACKBOARD IS 3/4 INCHES THICK BY 48 INCHES WIDE BY 72 INCHES LONG

Figure 6-1. Typical System 25 Equipment Area Floor Plan

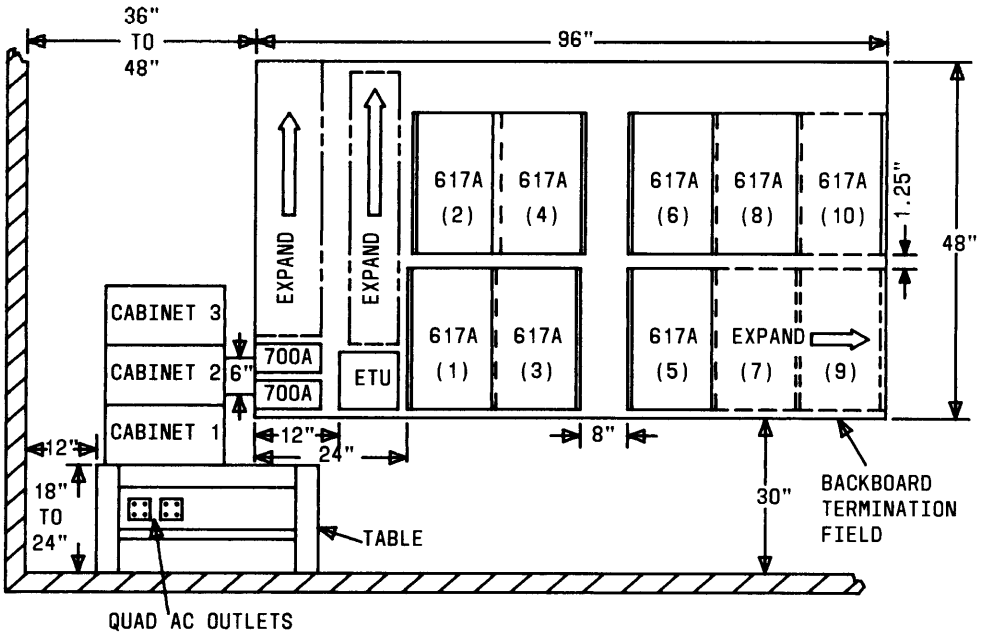


Figure 6-2. Typical System 25 Equipment Area Elevation Plan

Table Top Space

The following system equipment requires (customer provided) table top space in the equipment area:

- ***System Cabinets:*** Each cabinet is 13 inches high, 17 inches wide, and 21 inches deep. A 3-cabinet system requires a vertical space of approximately 40 inches and a 17-inch by 21-inch table top space. Each cabinet weighs approximately 75 pounds. Place the cabinets on a desk or table top that is about 18 inches high and capable of supporting at least 250 pounds. The cabinets *must* not be placed on the floor.

- ***System Administration Terminal (SAT) Model 703:*** The SAT should also be located near the system cabinets and plugged into the same AC outlet. It is 12 inches wide, 10 inches long, and 3 inches high.

- ***Digital Tape Unit:*** The Tape Unit (Model DC5 Digital Data Recorder) should also be located near the system cabinets. It is 5 inches wide, 2 inches high, and 10 inches long.

- ***SMDR or Call Accounting System (CAS):*** The AT&T Model 475 printer is approximately 16 inches wide, 12 inches long, and 6 inches high. The CAS runs on the AT&T PC 6300. The printer and the PC should also be located near the system cabinets.

Wall Space Requirements

The customer-provided backboard for the cross-connect field should be approximately 3/4 inch thick, 4 feet high, and 8 feet wide. Mount the board 30 inches above the floor. The board must conform to national and local fire safety codes.

If existing cross-connect hardware is reused, the space requirements and hardware requirements must be shown on the floor plan. Contact your AT&T Technical Consultant for assistance in planning for reuse of existing equipment.

Temperature and Humidity

The System 25 equipment should be installed in a well-ventilated area. The equipment *must* be located in an area with an ambient temperature between 40 degrees and 104 degrees Fahrenheit (5 and 40 degrees Celsius). The relative humidity must be less than 95%, noncondensing. These parameters should be maintained 24 hours a day, 7 days a week.

Air Purity

The cabinet should not be installed in an area where the air may be contaminated with any of the following:

- Excessive dust, lint, carbon particles, paper fiber contaminants, or metallic contaminants
- Contaminants expelled by office copying machines
- Highly corrosive atmosphere within an enclosed area or atmosphere containing vaporized chemical compounds that may condense on the equipment
- Explosive or flammable atmosphere.

Lighting

Lighting should be adequate to allow administration and maintenance personnel to perform their tasks. The recommended light intensity level is 50 to 70 footcandles. This level complies with the Occupational Safety and Health Act (OSHA) standards.

Electrical Noise/Radio-Frequency Interference (RFI)

In most cases, electrical noise is introduced to the system through trunk or voice terminal cables. However, electromagnetic fields near the system cabinets may also induce noise in the system. Therefore, the system cabinets and cable runs should not be placed in areas where a high electromagnetic field strength exists. Radio transmitters (AM or FM), television stations, induction heaters, motors (with commutators) of 0.25 horsepower (200 watts) or greater, and similar equipment are leading causes of interference. Small tools with universal motors are generally not a problem when they operate on separate power lines. Motors without commutators generally do not cause interference.

Field strengths below 1.0 volt per meter are unlikely to cause interference. Field strength can be measured by a tunable meter such as the Model R-70 meter manufactured by Electro-Metrics Division or broadband meters such as the HOLADAY* HI-3001 meter or Model EFS-1 meter manufactured by Instruments for Industry, Inc.

The field strength produced by radio transmitters can be estimated by dividing the square root of the emitted power in kilowatts by the distance from the antenna in kilometers. This yields the approximate field strength in volts per meter and is relatively accurate for distances greater than about half a wavelength (150 meters for a frequency of 1000 KHz).

* Trademark of Holaday Industries

AC Power Requirements

Each System 25 cabinet requires 500 Watts at 115V ac (maximum).

The System 25 power service must be a dedicated branch circuit with no other equipment served (see Figure 6-3). The customer should provide a load center of appropriate current rating (ITE EQ4 typical) equipped with 120V ac, 15 ampere (AMP), single pole magnetic circuit breaker(s) (ITE QP1-BO15 typical). Each breaker is to protect 2 associated wall mounted 115 V ac, 15 AMP, receptacles (HUBBELL 5262 typical). Grounding of this load center is to be provided by a “Green Wire” ground extended from the grounding electrode conductor at the AC service entrance to the load center.

The following materials are required:

- *Sing/e Cabinet System*

  1 - 15 AMP 3 Wire Dedicated Branch Service
  1 - 4"" Box (RACO 230 or Equiv.)
  1 - 4"" Cover (RACO 807 or Equiv.)
  1 - Ground Bar (Square D PK9GTA or Equiv.)
  2 - Recpt. (Hubbell 526215 AMP or Equiv.)

- *Multiple Cabinet System*

  2 - 15 AMP 3 Wire Dedicated Branch Service
  2 - 4"" Box (RACO 230 or Equiv.)
  2 - 4"" Cover (RACO 807 or Equiv.)
  1 - Ground Bar (Square D PK9GTA or Equiv.)
  4 - Recpt. (Hubbell 526215 AMP or Equiv.)

Typically, multiple cabinet systems can be powered from a single phase 120V ac, 60 Hertz service (two 15-amp circuits required). There are no phase restrictions between cabinets. Therefore, the two 15-amp circuits required may be derived from single or three-phase service.

The receptacles should be located at least 1 foot above the floor. Receptacles should not be located further than 4 feet from the cabinets.

DANGER

Under no circumstances should this equipment be connected to 220V ac; doing so poses a serious fire hazard.

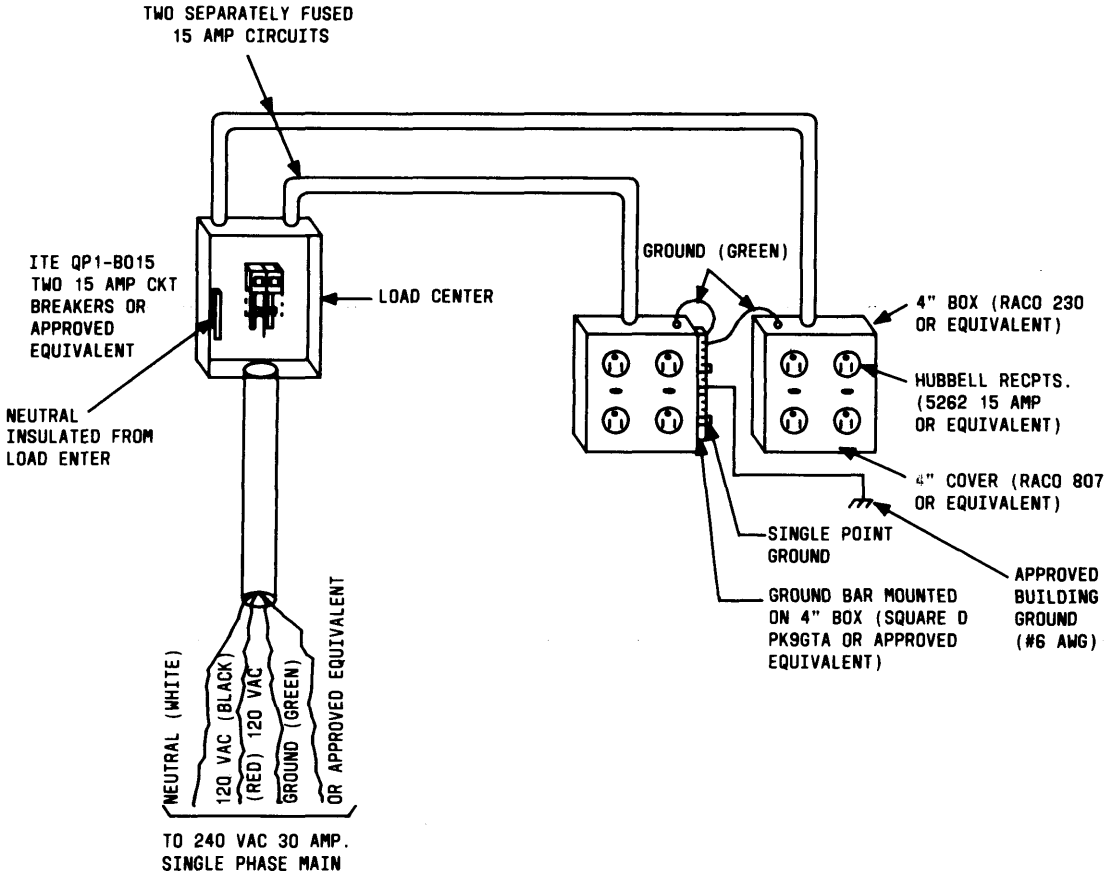


Figure 6-3. AC Power Distribution - Multiple Cabinet System

Grounding

Connection of an approved ground to the system cabinets is essential. An approved ground may consist of any of the following:

- Grounded Building Steel—The metal frame of the building.
- Water Pipe—A continuous metal water pipe, not less than 1/2 inch diameter, that is connected to an underground metal water pipe that is in direct contact with earth for 10 feet or more.
- Concrete-Encased Ground—An electrode encased by at least 2 inches of concrete and located within and near the bottom of a concrete foundation or footing in direct contact with the earth. The foundation must consist of at least 20 feet of one or more steel reinforcing bars or rods of not less than 1/2 inch diameter, or at least 20 feet of bare, solid copper wire no smaller than No. 4 gauge.
- Ground Ring—A ring that encircles a building or structure in direct contact with earth at a depth of at least 2-1/2 feet. The ring must consist of at least 20 feet of bare copper conductor no smaller than No. 2 gauge.

Lightning Protection

The System 25 lightning protections plan requires five distinct but interdependent items at each installation:

- Primary protection in the form of voltage limiters (typically carbon blocks or gas tubes). These devices bypass surges to approved building ground and limit potential differences between T/R pairs and building ground to less than 1500 volts.
- A single-point ground (SPG) system that connects green wire ground, System 25 system ground, and telephone company (TelCo) ground together and to approved building ground (Figure 6-3).
- Coupled bonding conductor (CBC) connected between the TelCo ground at the building entrance and System 25's SPG.

  CBC is tie-wrapped to all trunks to provide lightning protection; it can be a 16-gauge ground wire or continuous cable sheath. The CBC should also be run from the TelCo-provided network interface to the ground block of cabinet 1 (see Figure 3-4 in the "Functional Description" section). If the TelCo has not extended the CBC from the facility entrance to the network interface, the installer should run the CBC along the same route as the incoming facilities where feasible.
- Secondary protection provided within TN742 and all trunk CPs; such ports can withstand 800 volt metallic (differential) and 1500 volt longitudinal (common mode) surges.

- Surge protection on the AC power to System 25 and associated equipment (SAT, SMDR, DTU, etc.) provided by the Tll 428 Unit.

The protection outlined above is adequate for more than 99 percent of all lightning strikes. For the few remaining cases, external secondary protection, located at the trunk access area of the cross-connect field, can be employed. Several commercially available units can be used. If 66-type block terminations are used, a very convenient device is the LP3-230-220 Fused Lightning Protector manufactured by ITW-LINX, 195 Algonquin Road, Des Plaines, Illinois 60016. This unit plugs into the 66-block (in place of the shorting bars) and includes the sneak current fuse. One unit is required per protected pair. A ground bar is provided with the lightning protection units or can be ordered separately (Comcode 901007120). Porta Systems is another supplier of surge protection devices. Regardless of the type installed, all lightning protectors located in the cross-connect area must be grounded to System 25's SPG via a No. 6 AWG copper wire.

Transmission facilities (voice pairs) that extend out-of-building and are not included in the network interface may be installed by the TelCo or through other means (for example, by the building owner and/or the property owner). These facilities, aerial or buried, must be protected with voltage limiting devices and sneak current fuses. The only exception to this rule is buried cable less than 140 feet long that is enclosed in a continuous metallic conduit from one building entrance to the other. The conduit must be connected to an approved building ground at each end.

If single-line sets are extended out-of-building, primary protection devices for voice pairs should be used. If MERLIN Communications System voice terminals are extended out-of-building, the facilities must be protected with the Tll Model 341 (PEC 32918, Comcode 403865785) at each end. Terminals J1-3 and J1-6 of the Model 341 should not be connected. Maximum distance between a MERLIN System voice terminal and the System 25 cabinet is 2000 feet. There is no limit on how much of the 2000 feet can be out-of-building. However, out-of-building stations cannot be powered over the line; they must be powered locally using PEC 62510.

PARTS INFORMATION

Parts Listed by PEC 7-1

Parts Listed by COMCODE 7-9

PARTS INFORMATION

This section contains information that may assist you in cross referencing Apparatus Codes, Comcodes and Price Element Codes (PECs). The first table is grouped by descriptive Price Element Codes (PEC). The second table is arranged by component codes (COMCODE).

Parts Listed by PEC

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
6250-021	System 25 Control Unit (R2V1) Carrier	includes J58901A1 L4 (e/w)	
Part of:			
6250-021	Power Supply	WP90510	403954761
6250-021	Fans (2)	WP90677 L1	845416379
6250-021	Air Filter	21985-1	403957129
6250-021	Address Plug	*	845416635
6250-021	TDM Bus Term	ZHAF1	103810586
6250-021	CPU/MEM CP	ZTN129	105211031
6250-021	Service Circuit CP	ZTN85	103965323
6250-021	SLAC Graphics Overlays (2)	*	845875155
62501	System 25 Expansion Unit Carrier	includes J58901A1 L4 (e/w)	
Part of:			
62501	Power Supply	WP90510	403954761
62501	Fans (2)	WP90677 L1	845416379
62501	Air Filter	21985-1	403957129
62501	Address Plug	*	845416635
62501	TDM Bus Cable	J58901A4 L3	403961519
62502	GS Trunk CP	ZTN76	103965232
62503	LS Trunk CP	ZTN77	103965240
62504	TR Line CP	ZTN78	103965257
62505	ATL Line CP	ZTN79	103965265

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
62506	Asynchronous Data Unit	D181521 includes	105105506
Part of:			
62506	Mod Cord (14 ft)	D8W-87	103786802
62506	ATL ADU	Z3A5	103975349
62506	Power Supply	KS-22911 L1	403242639
62507	Digital Tape Unit	TS-555A	404079436
62508	System Administration Terminal (SAT)	TI-703	404079428
62509	Direct Extension Selector Console	23A1-003 e/w	103969424
Part of:			
62509	Mod Cord	D6AP-87	102937620
62509	Mod Cord (14ft)	D8W-87	103786802
62509	Power Supply	KS-22911 L1	403242639
62509	Adapter	400B	103848859
62510	MERLIN® Sys. Voice Terminal (VT) local power	D181522 includes	105105514
Part of:			
62510	Adapter	Z400F	103942857
62510	Mod Cord	D6AP-87	102937620
62510	Power Supply	KS-22911 L1	403242639
62511	Auxiliary Trunk Interface	D181523 includes	105105522
Part of:			
62511	Connecting Block	66E3-25	100009968
62511	Cable (15 ft DE)	B25A	100017334
62512	Paging/Dictation (Aux Trunk Interface)	D181524 includes	105105530
Part of:			
62512	Adapter	278A	103871844
62512	Power Supply	KS-22911 L1	403242639
62512	Mod/Spd Tip Cord (14ft)	D4BY	102999059
62512	Zener Kit	D181321	105031181

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
62513	Music-on-Hold Interface	D181575 includes	*
Part of:			
62513	Interface	KS-23395	405143186
62513	Transformer	2012D	102600517
62513	Voice Coupler	36A	103558916
62514	Emergency Transfer Unit	10B ETU	103984118
Part of:			
62514	Cables (2) (15 ft DE)	B25A	100017334
62514	Mod Cord (7ft)	D8W-87	103786786
62515	Peripheral Interface for On-Premises Direct or Switched Connection		
Part of:			
62515	Adapter	248B	102802113
62515	Adapter	355AF	105012645
62515	Adapter	400B2	104152558
62515	Mod Cord	D6AP-87	102937620
62515	ADU Crossover Cord	D8AM-87	104154430
62515	Mod Cords (2) (7 ft)	D8W-87	103786786
62515	EIA Crossover Cord	M7U-87	104246616
62515	ADU	Z3A4	103964185
62515	Transformer	2012D	102600517
62518	STARLAN Interface CP	ZTN84	103965315
62520	STARLAN/ATL Interface	D181807 includes	105355374
Part of:			
62520	Y-Adapter	WP90851 L1	405010620
62520	Adapter	KS-23475	405462904
62520	Power Supply	KS-22911 L1	403242639
62521	R2 Upgrade		
Part of:			
62521	CPU/MEM CP	ZTN129	105211031
62521	Documentation& Tapes for System 25 Upgrade	555-530-013	105528814
62521	SLAC Graphics Overlays(2)	*	845875155

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
62524	MERLIN Sys. Headset Adapter	502B	105471304
63111	Analog Line CP	TN742	103556957
63112	MET Line CP	TN735	103556882
63116	DID Trunk CP	TN753	103557062
63118	Aux Trunk CP	TN763	103557161
63119	Pooled Modem CP	TN758	103557112
63123	Tone Detector CP	TN748	103976163
63130	Data Line CP	TN726	103556791
63140	Tie Trunk CP	TN760B	103975645
21691	ADU Aux Power		
Part of:			
21691	Adapter	248B	102802113
21691	Adapter	400B	103848859
21691	Mod Cord	D6AP-87	102937620
21691	Transformer	2012D	102600517
1020-S90	ACCESS Software	*	105341218
1020-S91	CAM Software	*	105341382
1203-020, Rel 2	Advanced Admin. Software (MS-DOS)	*	105518039
1203-021, Rel 2	Advanced Admin. Software (UN IX)	*	105517941
1207-030	CMS Software	*	105517106
1205-010	VMS Software	*	105402283
2169-001	Tip Ring ADU	Z3A1	103963963
2169-002	Tip Ring ADU	Z3A2	*
2169-004	Tip Ring ADU	Z3A4	103964185

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
2169-005	Multiple ADU (MADU)	*	*
2301-SAA	Supplemental Alert Adapter	*	*
2403-004	Uninterruptible Power supply (UPS)	*	*
2610-001	STARLAN NETWORK Network Extension Unit (NEU)		527840003
2614-100	STARLAN NETWORK Network Access Unit (NAU)		527840102
2720-05P	25-pair/8-plug 15-ft Cable (Octopus Cable)	WP90780 L1	405010612
2720-05X	Splitter Cable, Tie-Trunk	WP90929 L3	403864150
2720-06X	Splitter Cable, CO Trunk	WP90929 L1	403836620
2724-29G	MADU Interface Cord	M48C	104109285
2724-30C	EIA Crossover Cord	M7U-87	104246616
2724-38X	ADU Crossover Cord	D8AM-87	104154430
2724-780	25-pair MADU Cable	*	*
2724-98G	MADU Interface Cord	M48G	104319025
2725-07G	Mod Cord (7 ft)	D8W-87	103786786
2725-075	Mod Cord (25 ft)	D8W-87	103786828
2750-A17	MET Adapter Cord	ZD8AJ	103881421
2750-A24	Mod/RS232 Adapt, Male	355A	105012637
2750-A25	Mod/RS232 Adapt, Female	355AF	105012645
2750-T05	Voice/Data (Y) Adapter	WP90851 L1	405010620

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
2781-004	System Wiring-Hourly Rate	*	*
2782-004	System Wiring-Flat Rate	*	*
2783-004	(New) System Wiring Run, Firm Quote	*	*
2788-004	System Wiring-Aftermarket, Flat Rate	*	*
2789-004	System Wiring-Aftermarket, Hourly Rate	*	*
3100-1TD	Basic TT Desk Tel	2500MMGB	*
3100-TRC	Basic TT Desk Tel with Recall Button	2500MMGT	*
3100-TWR	Basic TT Wall Tel	2554BM	103234472
3100-2TD	Basic TT Desk Tel (4A Speakerphone Compatible)	2500SM	*
3178-SYS	Basic TT Desk Tel with Message & Recall	2500DMGC	103966255
*	Basic TT Desk Tel with Headset Jack	2514 BMW	*
3100-ORD	Basic Rotary Desk Tel	500MM	103870267
3100-ORW	Basic Rotary Wall Tel	500BMPA	103823555
3100-2RD	Basic Rotary Desk Tel (4A Speakerphone Compatible)	500SM	103870416
3140-010	10-Button MET Set	2991C05	103871018
3141-BIS	10-Button MET Set with Built-in Speakerphone (BIS)	2993C04	103942146
Part of:			
3141-BIS	Transformer	2012D	102600517
3141-BIS	Kit of Parts	D181245	*

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
3143-12M	12-Buttom MET Set	Z7203M01A-003	103963310
3160-111	5-Buttom MERLIN Sys. VT (MAC30 Att)	Z7302H01D-003	*
3161-172	10-Button MERLIN Sys. VT (MAC30 Att)	Z7303H01D-003	*
3161-161	MERLIN Sys. HFAI VT	Z7309H01B-003	103982005
3162-412	34-But MERLIN Sys. VT	Z7305H01D-003	103842050
3162-417	34-But Dlx MERLIN Sys. VT (MAC30 Att)	Z7305H02D-003	103843538
3162-BIS	MERLIN Sys. BIS VT	Z7305H03B-003	103981965
3162-DIS	MERLIN Sys. VT with Display	Z7305H04C-003	103981981
3163-HFU	MERLIN Sys. Hands-Free Unit (Speakerphone)	S102A	103814356
3164-HFA	MERLIN Sys. Headset Adapter	502A	•
3165-10B	10-But MERLIN Sys. BIS VT (Black)	Z7313H01A-003	105336978
3166-22B	22-But MERLIN Sys. BIS VT (Black)	Z7314H01A-003	105336960
3167-34B	34-But MERLIN Sys. BIS VT (Black)	Z7316H01A-003	105336952
3167-DSB	34-But MERLIN Sys. BIS VT With Display (Black)	Z7317H01A-003	105400030
3170-OOM	Single-Line VT with Message Lamp & Recall Button	7101A01A-003	103871109
31710	Acoustic Coupler	349A Adapter	104010061

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
31017	Ground Start Key	55A1	100287085
31019	External Alert	*	*
31032	Message Waiting Indicator Adjunct	Z34A	•
•	CMS Voice Announcement Unit	*	405745811
	Modular Bulk Power Supply (includes)		
31760	Power Unit	346A	104174768
31761	Power Panel	346A-1	104174750
32918	IROB Unit	Tll Model 341	403865785
8310-001	AC Power Surge Suppressor	Tll Model 428	402988950

* Not Available.

Parts Listed by COMCODE

Throughout the following table, “part of” is abbreviated as “P/O.”

COMCODE PEC	DESCRIPTION	APPARATUS CODE
100009968 P/O 62511	Connecting Block	66E3-25
100017334 P/O 62514	Cables (2) (15 ft DE)	B25A
100017334 P/O 62511	Cable (15 ft DE)	B25A
100287085 31017	Ground Start Key	55A1
100963990 No PEC	Cable (15 ft DE)	A25D
102600517 P/O 21691 P/O 62515 P/O 3141-BIS P/O 62513	Transformer	2012D
102802113 P/O 21691 P/O 62515	Adapter	248B
102937620 P/O 62509 P/O 62510 P/O 62515 P/O 21691	Mod Cord	D6AP-87
102999059 P/O 62512	Mod/Spd Tip Cord (14ft)	D4BY
103104220 No PEC	Connecting Block	103A
103116943 NoPEC	Connecting Block	104A
103234472 3100-TWR	TT Wall Tel	2554BM
103556791 63130	Data Line CP	TN726
103556882 63112	MET Line CP	TN735
103556957 63111	Analog Line CP	TN742
103557062 63116	DID Trunk CP	TN753

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
103557112	63119	Pooled Modem CP	TN758
103557161	63118	Aux Trunk CP	TN763
103558916	P/O 62513	Voice Coupler	36A
103756334	No PEC	Connecting Block, Female	110
103786786	2925-07G P/O 62514 P/O 62515	Mod Cord (7ft)	D8W-87
103786802	2725-075 P/O 62506 P/O 62509	Mod Cord (14ft)	D8W-87
103786828	2725-075	Mod Cord (25ft)	D8W-87
103810586	P/O 6250-021	TDM Bus Term	J58901A1 L2,4
103814356	3163-HFU	MERLIN Sys. Hands-Free Unit (Speakerphone)	S102A
103823555	3100-ORW	Rotary Wall Tel	500BMPA
103842050	3162-412	34-Button MERLIN Sys. VT	Z7305H01B-003
103843538	3162-417	34-Button Deluxe MERLIN Sys. VT	Z7305H02B-003
103848859	P/O 21691 P/O 62509	Adapter	400B
103870267	3100-ORD	Rotary Desk Tel	500MM
103870416	3100-2RD	Rotary Desk Tel (4A Speakerphone Compatible)	500SM
103871018	3140-010	10-Button MET Set	2991C05

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
103871109	3170-00M	Single-Line VT with Message Lamp & RecalI Button	7101A01A-003
103871844	P/O 62512	Adapter	278A
103881421	2750-A17	MET Adapter Cord	ZD8AJ
103942146	3141-BIS	10-Button MET Set-BIS	2993C04
103942857	P/O 62510	Adapter	Z400F
103963310	3143-12M	12-Button MET Set	Z7203M01A-003
103963963	2169-001	Tip Ring ADU	Z3A1
103964185	2169-004 P/O 62515	Tip Ring ADU	Z3A4
103965232	62502	GS Trunk CP	ZTN76
103965240	62503	LS Trunk CP	ZTN77
103965257	62504	TR Line CP	ZTN78
103965265	62505	ATL Line CP	ZTN79
103965315	62518	STARLAN lnterface CP	ZTN84
103965323	P/O 6250-021	Service Circuit CP	ZTN85
103966255	3178-SYS	Basic TT Desk Tel with Message Lamp & Recall Button	2500DMGC
103969424	62509	Direct Extension Selector Console	23A1-003
103972907	No PEC	Mod to Mod Adapter	2210A1
103975349	P/O 62506	ATL ADU	Z3A5
103975645	63140	Tie Trunk CP	TN760B

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
103976163	63123	Tone Detector CP	TN748
103981965	3162-BIS	MERLIN Sys. BIS VT	Z7305H03B-003
103981981	3162-DIS	MERLIN Sys. Display VT	Z7305H04C-003
103982005	3161-161	MERLIN Sys. HFAl VT	Z7309H01B-003
103982658	No PEC	Panel (Part of SIP)	617A
103984118	62514	Emergency Trans Unit	10B ETU
104010061	31710	Adapter (Acoustic Coupler)	349A
104109285	2724-29G	MADU Interface Cord	M48C
104152558	P/O 62515	Adapter	400B2
104154430	2724-38X P/O 62515	ADU Crossover Cord	D8AM-87
104174750	31761	Power Panel	346A-1
104174768	31760	Power Unit	346A
104246616	2724-30C P/O 62515	EIA Crossover Cord	M7U-87
104319025	2724-98G	MADU Interface Cord	M48G
105012637	2750-A24	Mod/RS232 Adapt, Male	355A
105012645	2750-A25 P/O 62515	Mod/RS232Adapt, Female	355AF
105031181	P/O 62512	Zener Kit	D181321
105105506	62506	Asynchronous Data Unit	D181521
105105514	62510	MERLIN Sys. VT Local Power	D181522

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
105105522	62511	Auxiliary Trunk Interface	D181523
105105530	62512	Paging/Dictation	D181524
105196604	No PEC	Fanning Strip	50A
105211031	P/O 62521	CPU/MEM CP	ZTN129
105336952	3167-34B	34-But MERLIN Sys. BIS VT(Black)	Z7316H01A-003
105336960	3166-22B	22-But MERLIN Sys. BIS VT(Black)	Z7314H01A-003
105336978	3165-10B	10-But MERLIN Sys. BIS VT(Black)	Z7313H01A-003
105341218	1020-S90	ACCESS Software	*
105341382	1020-S91	CAM Software	*
105355374	62520	STARLAN/ATL Interface	D181807
105400030	3167-DSB	34-But MERLIN Sys. BIS VT With Display (Black)	Z7317H01A-003
105402283	1205-010	VMS Software	*
105471304	62524	MERLIN Sys. Headset Adapter	502B
105517106	1207-030	CMS Software	*
105517941	1203-021 Rel 2	Advanced Admin. Software (UNIX)	*
105518039	1203-020 Rel 2	Advanced Admin. Software (MS-DOS)	*
105528814	P/O 62521	Documentation& Tapes for System 25 Upgrade	555-530-013
402988950	8310-001	AC Power Surge Suppressor	Tll Model 428

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
403242639	P/O 62509 P/O 62510 P/O 62512 P/O 62520	Power Supply	KS-22911 L1
403836620	2720-06X	Splitter Cable, CO Trunk	WP90929L1
403864150	2720-05X	Splitter Cable, Tie-Trunk	WP90929L3
403865785	32918	lROB Unit	Tll Model 341
403954761	P/O 6250-021 P/O 62501	Power Supply	WP90510
403957129	P/O 6250-021 P/O 62501	Air Filter	21985-1
403961519	P/O 62501	TDM Bus Cable	J58901A1 L3
404079428	62508	System Admin Terminal (SAT)	TI-703
404079436	62507	Digital Tape Unit	TS-555A
405010612	2720-05P	25-pair/8-plug 15 ft Cable (Octopus Cable)	WP90780 L1
405010620	2750-T05 P/O 62520	Voice/Data (Y) Adapter	WP90851 L1
405177791	No PEC	110 to Mod Adapter	858A
405193186	P/O 62513	MOH Interface	KS-23395
405376377	No PEC	Connecting Block, Female	157BF
405462904	P/O 62520	STARLAN/ATL Interface Adapter	KS-23475 L1
405745811	•	CMS Voice Announcement Unit	*
527840003	2610-001	AT&T STARLAN NETWORK Network Extension Unit (NEU)	

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
527840102	2614-100	AT&T STARLAN NETWORK Network Access Unit (NAU)	
845412956	No PEC	CPU/MEM Interconnect Cable	*
845416379	P/O 6250-021 P/O 62501	Fans (2)	WP90677 L1
845416635	P/O 6250-021 P/O 62501	Address Plug	*
845875155	P/O 6250-021 P/O 62521	SLAC Graphics Overlays (2)	*

* Not Available.

REFERENCE DOCUMENTATION

Basic Manuals 8-1

Software Packages 8-1

Integrated Solution Documents 8-2

Descriptions of Basic Manuals 8-2

REFERENCE DOCUMENTATION

System 25 is supported by a complete set of basic and supplementary documentation and optional software. This section provides a brief summary of the available material for Release 2 Version 1 (R2V1).

Basic Manuals

• An Introduction to System 25	555-530-021
• R2V1 Administration Manual	555-530-500
• R2V1 Implementation Manual	555-530-650
• Installation and Maintenance Manual	555-530-103
• R2V1 New Capabilities Manual	555-530-205
• R2V1 Reference Manual	555-530-200
• R2V1 Terminal Operations Manual	555-530-710
• User Guides	
Data Features User Guide	555-520-704
Direct Trunk Attendant Console User Guide	555-530-701
Multiline Terminal User Guide	555-530-703
Single-Line Terminal User Guide	555-530-702
Switched Loop Attendant Console User Guide	555-530-706
• Remote Access Wallet Card	555-530-717

Software Packages

To expand System 25's communications and networking capabilities, AT&T offers the following optional software packages:

• R2V1 Advanced Administration Software Package (MS-DOS)	PEC 1203-020, Release 2
• R2V1 Advanced Administration Software Package (UNIX)	PEC 1203-021, Release 2
• Call Management System (Includes next four documents)	PEC 1207-030
Call Management System Agent Card	
Call Management System Installation & Startup Guide	
Call Management System Planning Guide	
Call Management System System Manual	
• Communications Access Manager Software Package	PEC 1020-S91
• STAR LAN NETWORK ACCESS Software Package	PEC 1020-S90

Integrated Solution Documents

- Integrated Solution User Guide 555-530-715
- Integrated Solution Instructor's Guide 555-530-717
- Integrated Solution Student Guide 555-530-718
- AT&T Call Accounting System Reports Guide 775-413

Descriptions of Basic Manuals

An Introduction to System 25

Provides a summary of System 25 features, services, and equipment in an attractive full-color format with many pictures. The emphasis is on how System 25 helps solve information management, productivity, and cost control problems.

Administration Manual

Provides the information necessary to initialize a system and to perform on-going system administration. Explains the operation of the System Administration Terminal, the Digital Tape Unit, and the commands that allow the System Administrator to make changes and additions.

Implementation Manual

Describes how to plan the operating configuration of the system. Explains how to determine customer needs and how to convert these needs into a system configuration plan. This plan is recorded on accompanying forms that are used in conjunction with the Administration Manual to initialize the system. The Implementation Manual and associated forms are packaged together in the Administration Records Binder.

Installation And Maintenance Manual

Provides step-by-step procedures for installing System 25 and associated equipment and procedures for isolating and clearing customer affecting faults. Includes procedures for testing equipment and trunks and for making additions and changes to the system.

New Capabilities Manual

Describes the System 25 Release 2 Version 1 (R2V1) features that were not included in Release 1 Version 2 (R1V2). In addition, it covers feature enhancements and new hardware.

Reference Manual

The principal technical reference for users of System 25. It provides reference material for sales support, system configuration and operation and for the system technician. It contains a comprehensive description of the system, emphasizing features, components and overall capabilities and capacities.

Terminal Operations Manual

Designed to help the System Administrator better understand System 25 voice terminal and data terminal operation. This manual contains the operating procedures provided in the various User Guides and provides additional explanatory material as well.

User Guides (700 Series)

Contain step-by-step operating procedures for System 25 attendants and users of voice and data terminals.

GLOSSARY

This section provides explanations for acronyms and definitions of terms used in this manual.

ADU: (Asynchronous Data Unit)

ARS: (Automatic Route Selection)

ASCII: (American Standard Code for Information Exchange)

Administer

To access and change the parameters associated with the services or features of the system.

Analog Data Endpoint

Data endpoints with customer provided built-in or stand-alone modems. They do not require the use of data modules (asynchronous data units) and are addressed similar to any voice terminal by PDC. These end-points connect to tip/ring type circuit pack ports.

Analog Voice Terminals

Voice terminals served by a single-line tip and ring circuit (2500 series and 7101A voice terminals or industry standard Dual Tone Multifrequency equivalent).

Appearance

See Call Appearance.

Asynchronous Data Transmission

A scheme for transmitting data where each character is preceded by a start bit and followed by a stop bit, thus permitting data elements to occur at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard).

Asynchronous Data Unit (ADU)

A data communications equipment (DCE) type device that allows direct connection between RS-232C equipment and the digital switch via ports on the Data Line Circuit Pack (TN-726).

Attendant

The operator of the attendant console.

Attendant Console

- **Direct Trunk or Switched Loop Attendant Console:** An electronic call-handling position with pushbutton control. Used by attendants to answer and place calls and to manage and monitor some system operations.
- **Direct Extension Selector Console:** Provides the attendant with a visual indication of the active or idle status of extension numbers assigned in the system. Also allows the attendant to extend calls to system users by operation of appropriate Hundreds Group Select buttons and associated Direct Extension Selection (DXS) buttons.

Auxiliary Equipment

- Dictation Equipment
- Delay Announcement Devices
- External Alerting Devices (external alerts)
- Music-On-Hold Equipment
- Paging Equipment

Auxiliary Trunk

A trunk circuit used to connect auxiliary equipment to the switch, for example, music or dictation equipment.

BLF: (Busy Lamp Field)

BPS: (Bits Per Second)

Bit (Binary Digit)

One unit of information in binary notation (having two possible states or values, zero or one).

Bridge (Bridging)

The sharing of the same extension or line by two or more voice terminals.

Buffer

A circuit or component that isolates one electrical circuit from another. Typically, a buffer holds data from one circuit or process until another circuit or process is ready to accept the data.

Bus

A multi-conductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

Bus, Time Division Multiplex

See Time Division Multiplex Bus.

CAM: (Communications Access Manager)

CAS: (Call Accounting System)

CMS: (Call Management System)

CCS (Hundred Call Seconds)

A traffic-measuring unit that expresses the load of one or more traffic-handling devices. A device used for 1 hour without interruption generates 36 CCS which equals 1 erlang (see Erlang).

Call Appearance, Voice Terminal

A button (for example, System Access, Bridged Access, Loop, DSS, Flex DSS, or Auto Intercom) used to place outgoing calls, receive incoming calls, or hold calls. Two LEDs next to the button show the status of the call appearance or status of the call.

Central Office (CO)

The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

Central Office Exchanges

The first three digits of a 7-digit public network telephone number. These codes are numbered from 200 through 999 and are sometimes referred to as NNXs.

Central Office Trunk

A telecommunications channel that provides access from the system to the public network through the local central office.

Channel

A communications path for transmitting voice and data.

Class of Service (COS)

Parameters used to define voice terminal, data, Remote Access, and trunk port capabilities and restrictions.

Common Control Switching Arrangement (CCSA)

A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

Confirmation Tone

Three short bursts of tone followed by silence; indicates that the feature activated, deactivated, or canceled has been accepted.

Console

See Attendant Console.

Coverage Call

A call that is redirected from the called party's personal dial code to an alternate answering position when certain criteria are met.

Coverage Path

The order in which calls are redirected to alternate answering positions.

Coverage Point

The attendant positions (as a group), Direct Group Call (DGC) group, Coverage Receiver Group, or a voice terminal extension designated as an alternate answering position in a coverage path.

Covering User

The person at an alternate answering position who answers a redirected call (the coverage receiver).

DCE: (Data Communications Equipment)

DDC: (Data Dial Code)

DDD: (Direct Distance Dialing)

DID: (Direct Inward Dialing)

DGC: (Direct Group Calling)

DTAC: (Direct Trunk Attendant Console)

DTE: (Data Terminal Equipment)

DTU: (Digital Tape Unit)

DTMF: (Dual Tone Multifrequency)

DXS: (Direct Extension Selector)

Data Channel

A communications path between two points used to transmit digital signals.

Data Communications Equipment (DCE)

Refers to a specific RS-232C interface connector configuration. DCE devices are designed to interface directly (pin-for-pin) to Data Terminal Equipment (DTE). The transmit and receive pins are reversed from that of a DTE interface. A modem is an example of a DCE device.

Data End Point

Two general groups; those having a DTE type interface, which encompasses almost all of the data terminal devices, and the group of DCE interface devices which are primarily modems. However, it must be noted that within each category, control interfaces may also vary. Refer to Analog Data Endpoint and Digital Data Endpoints for additional information.

Data Module

A data interface device (i.e., Asynchronous Data Unit) providing a standard interface between customer-provided data equipment and a data port on the switch.

Data Terminal Equipment (DTE)

DTE refers to a specific RS-232C connector termination designed to connect directly to a DCE type connection. Typically associated with video display terminals, printers, and computers which either originate or terminate a data transmission path.

Data Terminals

Refers to RS-232C compatible Data Terminal Equipment.

Delay-Dial Tie Trunk

After a request for service (called a seizure) is detected on an incoming trunk, the system sends a momentary signal followed by a steady tone over the trunk. This informs the calling party that dialing can start. This type of trunk allows dialing directly into the system. That is, the digits are received as they are dialed.

Digital Data Endpoints

In System 25, digital data endpoints include any digital device providing a RS-232C connection interface to the switch. The connection is via Asynchronous Data Units (ADUs) to the switch.

Direct Extension Selector (DXS) Console

An option at the attendant console that allows an attendant direct access to voice terminals by pressing a Group Select button and a DXS button.

EIA: (Electronics Industries Association)

Emergency Transfer Unit

Provides direct connection of designated Power Failure Transfer (PFT) registered voice terminals to the CO during a power failure or other service interruption.

Erlang

A traffic measuring unit that expresses the load of one or more traffic-handling devices [36 CCS equals 1 erlang - see CCS (Hundred Call Seconds)].

Extension Number

One- through four-digit number assigned to each voice terminal and data end point in the system. Also see "Personal Dial Code"

Extended Station

A single-line voice terminal located more than 2000 feet from the system cabinets.

External Call

A connection between a system user and a party on the public telephone network or on a tie trunk; usually referred to as an outside or trunk call.

FRL: (Facility Restriction Level)

Facility (physical)

A transmission channel to another switching system; to a Central Office for example. By application, examples are:

- CO Trunks
- FX Trunks
- WATS Trunks
- Tie Trunks

By technical type these include loop start, ground start, DID, automatic ringdown, etc. These facilities may be accessed by their facility access codes (FACs).

Feature

A specifically defined function or service provided by the system.

Feature Button

A labeled button on a voice terminal or attendant console designating a specific feature.

Foreign Exchange (FX)

A central office other than the one providing local access to the public telephone network.

Foreign Exchange Trunk

A telecommunications channel that directly connects the system to a central office other than its local central office.

Foreign Numbering Plan Area Code (FNPA)

An area code other than the local area code; also known as the “other area code.” The foreign area code must be dialed to call outside the local geographical area.

Ground-Start Trunk

On outgoing calls, System 25 transmits a request for services to the distant switching system by grounding the trunk ring lead. When the distant system is ready to receive the digits of the called number, that system grounds the trunk tip lead. When the System 25 detects this ground, the digits are sent. (Tip and ring are common nomenclature to differentiate between ground-start trunk leads.) On incoming calls, detection of ground on the tip lead is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

HFAI: (Hands-Free Answer on Intercom)

Home Numbering Plan Area Code (HNPA)

The local area code; also known as the “home area code.” The local area code does not have to be dialed to call numbers within the local geographical area.

Immediate-Start Tie Trunk

After establishing a connection with the distant switching system for an outgoing call, the system waits a nominal 65 milliseconds before sending the digits of the called number. This allows time for the distant system to prepare to receive the digits. Similarly, on an incoming call, the system has less than 65 milliseconds to prepare to receive the digits.

Inside Call

A connection between two parties within the system.

Intercept Tone

An alternating high and low tone; indicates a dialing error or denial of the service requested.

Interface

A common boundary between two systems or pieces of equipment.

Internal Call

A connection between two users within the system. Same as an inside call.

I-Use LED

A red LED on a multiline voice terminal that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

LDN: (Listed Directory Number)

LED: (Light Emitting Diode)

Loop Start Trunk

After establishing a connection with the distant switching system for an outgoing call, System 25 waits for a short period of time before sending the digits of the called number. On incoming calls, the received request for service is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

MET: (Multibutton Electronic Telephone)

Modem

A device that modulates and demodulates signals transmitted over a communications path. Used to connect Data Terminal Equipment to the system's analog ports. The system provides a pooled modem conversion resource (12 resources maximum per system—212A compatible).

Modem Pooling

Provides shared-use conversion resources that eliminate the need for a dedicated modem when an analog data end point accesses, or is accessed by, an analog line or trunk.

Multiline Voice Terminal

A terminal equipped with several call appearance buttons for the same extension number. Allows the user to handle more than one call, on that same extension number, at the same time.

Multiplexed

The simultaneous transmission of two or more signals over a common transmission medium.

NPA: (Number Plan Area)

Network

An arrangement of inter and/or intra location circuits designed to perform specific functions.

Network Interface

Provided by the CO telephone company in two forms:
(1) RJ21X for trunk facilities other than tie trunks.
(2) RJ2GX for tie trunk facilities.

Off-Premises Station (OPS)

An arrangement provided by the local telephone company which permits remote Terminal Equipment to operate as though it was directly connected to the System 25. This tariffed service can only be provided for FCC registered single-line voice terminals.

Out-Of-Building Station

The Terminal Equipment is directly connected to the System 25, but is not located in the same building as the common equipment. Special arrangements are made to protect the system and its users from lightning, power line crosses, etc. Only the single-line and 7300H series of voice terminal may be so connected. MET Sets can not be connected as Out-Of-Building stations.

Outside Call

A connection between a system user and a party on the public telephone network or on a tie trunk.

PDC: (Personal Dial Code)

PFT: (Power Failure Transfer)

Paging Trunk

A telecommunications channel used to access an amplifier for loudspeaker paging.

Parameter

Any set of physical properties whose values determine the characteristics or performance of a system.

Peripheral Equipment

System Administration Terminal (SAT), SMDR Output device such as a SMDR Printer or a Call Accounting System, Digital Tape Unit (DTU).

Personal Dial Code

Each system user is assigned a PDC and is allowed to “sign in” the PDC at any voice terminal in the system as he or she moves about the premises. The PDC may be a 1-, 2-, 3-, or 4-digit number. There are two types of PDCs:

- PDCs assigned to voice terminals - Associated with each voice terminal in the system.
- Floating PDCs (FPDCs) - Assigned to visitors and those users who do not have exclusive use of a voice terminal. An FPDC may be signed in by its owner at any system voice terminal. Calls to the FPDC will ring at the terminal where it is signed in. Calls to an FPDC that is not signed in anywhere will either receive reorder tone or be directed to the attendant (administrable).

Pickup Group

A group of individuals authorized to answer any call directed to an extension number within the group.

Port

An interface circuit between System 25 and associated auxiliary and peripheral equipment. Typical references include:

- Terminal port (station port)
- Facility port (trunk port)
- Auxiliary equipment port

Private Branch Exchange (PBX)

A switching system that provides switched communications access amongst its terminals and facilities (e.g., System 25)

Private Network

A network used exclusively for handling the telecommunications needs of a particular customer.

Private Network Office Code (RNX)

The first three digits of a 7-digit private network number. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

Protocol

A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

Public Network

The network that can be openly accessed by all customers for local or long-distance calling.

Queue

An ordered sequence of calls waiting to be processed.

Queuing

The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, to a station, or to a trunk. Calls are automatically connected in first-in, first-out sequence.

RIMS: (Remote Initialization and Maintenance Service)

Random Access Memory (RAM)

A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

Read Only Memory (ROM)

A storage arrangement primarily for information retrieval applications.

Recall Dial Tone

Three short bursts of tone followed by steady dial tone; indicates the system has completed some action (such as holding a call) and is ready to accept dialing.

Redirection Criteria

The information administered for each voice terminal that determines when an incoming call is redirected to coverage.

Reorder Tone

A fast-busy tone repeated 120 times a minute; indicates that a call attempt cannot be completed because, for example, all trunks are busy, a dialing error has occured, the terminal is restricted from making the call, or a required account code was not entered.

SAT: (System Administration Terminal)

SIP: (Station Interconnect Panel)

SLAC: (Switched Loop Attendant Console)

SMDR: (Station Message Detail Recording)

Selector Console: (Direct Extension Selector Console)

Single-Line Voice Terminal

Voice terminal served by a single-line tip and ring circuit (2500 series and 7101 A voice terminals or industry standard Dual Tone Multifrequency equivalent).

Software

A set of computer programs that accomplish one or more tasks.

Split

A condition whereby a caller is temporarily separated from a connection with the attendant. This split condition automatically occurs when the attendant, active on a call, either presses the Start button or uses the Direct Extension Selector Console.

Status LED

A green LED that shows the status of a call appearance or a feature button by the state of the lamp (lighted, flashing, fluttering, broken flutter, or dark).

Station

A place where terminal equipment is located or sometimes the terminal equipment itself. Each voice terminal (station) is assigned a station (extension) number. Users of the terminal are sometimes referred to as station users. Reference to the extension number is usually in the form PDC (Personal Dial Code) rather than station number. Though PDCs may be "signed-in" at other stations, in most discussions PDCs and station numbers are interchangeable. Analogously, data stations are assigned DDCs (Data Dial Codes).

Switch

The software-controlled communications processor complex that interprets dialing pulses/tones/keyboard characters and makes the proper interconnections both within the system and external to the system. The switch itself consists of a digital computer, software, storage device (memory), and associated circuit packs and special hardware necessary to perform the actual connections.

Switchhook

The button(s) on a voice terminal located under the handset.

System Administrator

A person responsible for specifying and administering features and services for the system.

System Restore

A process that allows stored data to be written from a tape or PC file into the system memory.

TAE: (Trunk Access Equipment)

Terminal Equipment:

Equipment for changing information (sound, keystrokes) into an electrical signal compatible with the system's port circuits (voice and data terminals are two subdivisions).

Tie Trunk

A telecommunications channel that directly connects two private switching systems.

Time Division Multiplex Bus

A special bus that is time shared by preallocating short time slots to each transmitter on a regular basis. In a PBX, all port circuits are connected to the time division multiplex bus permitting any port to send a signal to any other port.

Tone Ringer

A device with a speaker, used in electronic voice terminals to alert the user.

Translations

Specific information assigned to a terminal or to the system and customized for the user.

Trunk

A telecommunications channel between two switching systems.

Trunk Group

Telecommunications channels assigned as a group for certain functions.

Trunk Port

The hardware providing the access point to the system switching network for each circuit associated with a trunk.

Unit Load

A measurement used to evaluate a System 25 cabinet's power load capacity. Each System 25 cabinet can handle 80 unit loads of 48-volt power. One unit load equals 44mA.

Voice Terminal

A single-line or multiline voice instrument; a telephone.

Voice Terminal Adjuncts

Devices that can be connected to voice terminals to provide additional services (headset adapters, speakerphones, acoustic couplers, etc.).

Wide Area Telecommunications Service (WATS)

A service that allows calls to a certain area or areas for a flat-rate charge based on usage.

Wink-Start Tie Trunk

After establishing a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

Write Operation

The process of putting information onto a storage medium such as magnetic tape.

800 Service

A service that allows incoming calls from a certain area or areas to an assigned number for a flat-rate charge based on usage.

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